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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **28 Apr 09**

MEMORANDUM FOR DISTRIBUTION

SUBJECT: Special Interoperability Test Certification of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7

References: (a) DoD Directive 4630.5, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01D, "Interoperability and Supportability of Information Technology and National Security Systems," 8 March 2006
(c) through (f), see Enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Cisco Unified CallManager Version 4.3(2) SR1b with IOS Software Release 12.4(15) T7 is hereinafter referred to as the system under test (SUT). The SUT meets all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) for the following switch types: Private Branch Exchange (PBX) 1 and PBX 2. The SUT meets the Voice over Internet Protocol critical interoperability requirements with any certified Assured Services Local Area Network (ASLAN) on the Unified Capabilities (UC) Approved Products List (APL). The identified test discrepancies shown in the Certification Testing Summary (Enclosure 2) have an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC, or authorized by the Program Management Office for use within the DSN. This certification expires upon changes that could affect interoperability, but no later than three years from the date of this memorandum.

3. This finding is based on interoperability testing conducted by JITC, DISA adjudication of open test discrepancy reports, review of the vendor's Letters of Compliance (LoC), and Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation. Interoperability testing of the SUT was conducted at JITC's Global Information Grid Network Test Facility at Fort Huachuca, Arizona, from 3 November through 19 December 2008. DISA adjudication of outstanding test discrepancy reports and review of the vendor's LoC was completed on 17 December 2008. DSAWG grants accreditation based on the security testing completed by DISA-led Information Assurance test teams and published in a separate report

(reference (c)). DSAWG accreditation was granted on 21 April 2009. Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The SUT certified hardware and software components are listed in Table 1. The interoperability test summary of the SUT is indicated in Table 2. The PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 3. This interoperability test status is based on the PBX 1's ability to meet:

a. DSN services for Network and Applications specified in reference (d).

b. PBX 1 interface and signaling requirements for trunks/lines specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

c. PBX 1 CRs/FRs specified in reference (e) verified through JITC testing and/or vendor submission of LoC.

d. The overall system interoperability performance derived from test procedures listed in reference (f).

Table 1. SUT Hardware and Software Components

Cisco Unified CallManager Version 4.3(2) SR1b , with IOS Software Release 12.4(15) T7			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
CallManagers **MCS7835H2**, **MCS7825H3**, **MCS7825H2**, **MCS7835H1**, MCS7835H, MCS7835I1, MCS7845H2, MCS7825H, MCS7835I, MCS7845H, MCS7845I, MCS7825-H1, MCS7825I1, MCS7845H1, MCS7845I1	4.3(2) SR1b	Not Applicable	Processing/Signaling
Cisco 3745/3725 Multiservice Access Router (Gateway) (See note 2.)	IOS 12.4(15) T7	**NM HD 2V**	TDM Interface NM HD Voice, 2-slot IP communications voice/fax
		NM HD 2VE	TDM Interface NM HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		VWIC 1MFT T1	Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1
		VWIC 2MFT T1	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1
		VWIC 2MFT T1 DI	Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1, Drop and Insert
Cisco 3845/3825 Integrated Services Router (Gateway)	IOS 12.4(15) T7	**NM HDV2**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 3.)

Table 1. SUT Hardware and Software Components (continued)

Cisco Unified CallManager Version 4.3(2) SR1b with IOS Software Release 12.4(15) T7 (continued)			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Cisco 2851 Integrated Services Router (Gateway)	IOS 12.4(15) T7	NM HDV2	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G (See note 2.)	Load: P00308000900	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	Load: SCCP70.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7911G and CP-7906G	Load: SCCP11.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	Load: SCCP41.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	Load: SCCP42.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	Load: SCCP45.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	Load: SCCP75.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000300	Not Applicable	Expansion module
7915	B015-1-0-2	Not Applicable	Expansion module
7916	B016-1-0-2SR1	Not Applicable	Expansion module
CIS 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	CP-7961G IP phone, TEMPEST version
CIS 7975G (See note 5.)	SCCP75.8-4-1SR1S	Not Applicable	7975G IP phone, TEMPEST version
CRYPTEK 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	7961G IP phone, TEMPEST version with no PC interface and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones

Table 1. SUT Hardware and Software Components (continued)

NOTES:

1 Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same IOS software and hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

2 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.

b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.

c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

3 These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.

4 The EM HDA 8FXS expansion module requires the EVM HD module. Up to two EM HDA 8FXS expansion modules are supported for each EVM HD.

5 CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.

LEGEND:

10/100BaseT	10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet	GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)	PSTN	Public Switched Telephone Network
		HD	High Density	RJ	Registered Jack
CP	Cisco Phone	HDA	High Density Analog	SCCP	Skinny Client Control Protocol
DI	Drop and Insert	IOS	Internetwork Operating System	SR	Service Release
DID	Direct Inward Dialing	IP	Internet Protocol	SUT	System Under Test
DSN	Defense Switched Network	IPv6	Internet Protocol version 6	T1	Digital Transmission Link Level 1 (1.544 Mbps)
E1	European Basic Multiplex Rate (2.048 Mbps)	ISDN	Integrated Services Digital Network		
		JITC	Joint Interoperability Test Command	TDM	Time Division Multiplexing
EM	Expansion Module	Mbps	Megabits per second	UCR	Unified Capabilities Requirements
EVM	Extension Voice Module	MCS	Media Convergence Server		
Fax	facsimile	MFT	Multiflex Trunk	V	Voice
FXS	Foreign Exchange Station	ms	milliseconds	VE	Voice/Fax Enhanced
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)	NM	Network Module	VIC	Voice Interface Card
		PC	Personal Computer	VWIC	Voice WAN Interface Card
		PRI	Primary Rate Interface	WAN	Wide Area Network

Table 2. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release and have not been fixed by the vendor. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA.[4] When channels on a T1 CAS trunk group are busied by the remote switching system, the SUT fails to acknowledge all of these busy outs. As a result calls originated by the SUT fail to complete and the proper treatment is not provided. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA.[4] This interface does not support NFAS.[5] The SUT monitoring tool occasionally provides inaccurate reports when a remote trunk is busy.[6]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with a minor configuration change[7] and the following minor exceptions: The SUT does not support an MLPP global diversion number.[8] Diversion to the alternate directory number is provided in lieu of the BNEA.[9] When an analog line is preempted at a precedence higher than the already established call, the analog interface will ring at ROUTINE.[10] The SUT gateway analog interface does not support required line features.[11] The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
VoIP	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number.[8] Diversion to the alternate directory number is provided in lieu of the BNEA.[9] The operational impact is minor.
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: Full compliance of DSN Common Call Features was not met.[11, 12, 13, 14, 15, 16, 17] The operational impact is minor.
Attendant	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Public Safety	Yes	Certified	All public safety features are conditional. The SUT Met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features.[18]

Table 2. SUT Interoperability Test Summary (continued)

DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Conferencing	Yes	Certified	Meet-Me Conferencing is met through the use of the Cisco MeetingPlace. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the Loss of Command and Control announcement.[19]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Certified	Met all critical CRs and FRs.[5]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	Certified	See note 20.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN posted on the UC APL. See notes 21 and 22.

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: This interface does not support NFAS.[5] The operational impact is minor.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.

NOTES:

1 T1 CAS wink start recognition is not within the required tolerance of 100 ms to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

2 The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

3 During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

4 ROUTINE calls attempted over a trunk that is broken receive a T120 in lieu of an ICA. Calls above ROUTINE attempted over a trunk that is broken receive a BPA in lieu of an ICA. The operational impact is minor because they are treated with a BPA and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

5 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

Table 2. SUT Interoperability Test Summary (continued)

NOTES (continued):

6 A discrepancy exists that is associated with the monitoring tool that SUT uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

7 A configuration change was required on the analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

8 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified CallManager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

9 When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the BNEA is not provided. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.

10 When an analog station is active with a call and is preempted by a higher precedence call, the analog station receives the proper PNT. However, after going on hook, the station rings at ROUTINE. This was found to be a minor impact because the station is still preempted correctly.

11 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX 1 for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a PBX 1.

12 The SUT does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

13 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

14 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

15 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

16 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. DISA adjudicated this anomaly as having a minor operational impact because the preempted user receives PNT and the other members remain connected.

17 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for Precedence above ROUTINE calls. There is no operational impact.

18 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

19 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

20 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

21 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

Table 2. SUT Interoperability Test Summary (continued)

NOTES (continued):

22 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.
b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.
c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

LEGEND:

ANSI	American National Standards Institute	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
APL	Approved Products List		
ASLAN	Assured Services Local Area Network	Mbps	Megabits per second
BNEA	Busy Not Equipped Announcement	MFR1	Multi-Frequency Recommendation 1
BPA	Block Precedence Announcement	MLPP	Multi-Level Precedence and Preemption
BRI	Basic Rate Interface	ms	milliseconds
C2	Command and Control	NI 1/2	National ISDN Standard 1 or 2
CAS	Channel Associated Signaling	NI2	National ISDN Standard 2
CRs	Capability Requirements	NFAS	Non Facility Associated Signaling
DISA	Defense Information Systems Agency	PBX 1	Private Branch Exchange 1
DP	Dial Pulse	PMO	Program Management Office
DSN	Defense Switched Network	PNT	Preemption Notification Tone
DSS1	Digital Subscriber Signaling 1	PRI	Primary Rate Interface
DTMF	Dual Tone Multi-Frequency	PSTN	Public Switched Telephone Network
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.931	Signaling Standard for ISDN
FRs	Feature Requirements	Q.955.3	ISDN Signaling standard for E1 MLPP
GR	Generic Requirement	SS7	Signaling System 7
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	SUT	System Under Test
ICA	Isolated Code Announcement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
IPv4	Internet Protocol version 4	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IPv6	Internet Protocol version 6		
ISDN	Integrated Services Digital Network	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	UC	Unified Capabilities
		UCR	Unified Capabilities Requirements
JITC	Joint Interoperability Test Command	VoIP	Voice over Internet Protocol

Table 3. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
			• PBX Line (C)	• UCR Section 2.3.1
			• Direct Inward Dialing (C)	• UCR Section 2.3.2
			• National ISDN 1/2 Primary Access (R)	• UCR Section 2.3.4.1
			• ISDN ANSI MLPP Service Capability (R)	• UCR Section 2.3.4.1.1
			• ITU-T ISDN Primary Access (Europe only) (C)	• UCR Section 2.3.4.2
			• ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C)	• UCR Section 2.3.4.2.1
			• Normal Wink Start Operations (C)	• UCR Section 5.3.3.1.1
T1 CAS (MFR1, DTMF, DP)	No		• Glare Operation (C)	• UCR Section 5.3.3.1.2
			• Abnormal Wink Start (C)	• UCR Section 5.3.3.2.1
			• Glare Resolution (C)	• UCR Section 5.3.3.2.2
			• Call for Service Timing (R)	• UCR Section 5.3.5
			• Guard Timing (R)	• UCR Section 5.3.6
			• Satellite Timing (C)	• UCR Section 5.3.7
			• Disconnect Control (C)	• UCR Section 5.3.8
			• Reselect and Retrial (C)	• UCR Section 5.3.9
			• Off-Hook Supervision Transition (C)	• UCR Section 5.3.10
E1 CAS (MFR1, DTMF, DP)	No (Europe only)		• Dial-Pulse Signals (C)	• UCR Section 5.4.1
			• DTMF Signaling (C)	• UCR Section 5.4.2
			• Standard Digit Format for Precedence (C)	• UCR Section 5.4.2.1
			• MFR1 2/6 Signaling (C)	• UCR Section 5.4.3
			• Alerting Signals and Tones (R)	• UCR Section 5.5
			• DSN ISDN User-to-Network Signaling (R)	• UCR Section 5.7.1
		Trunking	• Application (R)	• UCR Section 5.7.1.1
			• Physical Layer (R)	• UCR Section 5.7.1.2
			• Data Link Layer (R)	• UCR Section 5.7.1.3
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes		• Data Link Connection (R)	• UCR Section 5.7.1.3.1
			• Peer-to-Peer Procedures of Data-Link Layer (R)	• UCR Section 5.7.1.3.2
			• Layer 3 DSN User-to-Network Signaling (R)	• UCR Section 5.7.1.4
			• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	• UCR Section 5.7.1.4.2
			• Sequence of Messages for DSN Circuit-Switched Calls (R)	• UCR Section 5.7.1.4.3
			• Message Functional Definition and Content (R)	• UCR Section 5.7.1.4.4
			• General Message Format and Information Elements Coding (R)	• UCR Section 5.7.1.4.5
			• Supplementary Services (C)	• UCR Section 5.7.1.4.6
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)		• PCM-24 Digital Trunk Interface (R)	• UCR Section 7.1
			• Interface Characteristics (R)	• UCR Section 7.1.1
			• Supervisory Channel Associated Signaling (C)	• UCR Section 7.1.2
			• Clear Channel Capability (R)	• UCR Section 7.1.3
			• Alarm and Restoral Requirements (R)	• UCR Section 7.1.4
			• PCM-30 Digital Trunk Interface (Europe only) (C)	• UCR Section 7.2
			• Interoperation of PCM-24 and PCM-30 (C)	• UCR Section 7.3
			• Analog Trunk Interface (C)	• UCR Section 7.4
			• Integrated Digital Loop Carrier (C)	• UCR Section 7.5
			• Trunk Group-Remove from Service (C)	• UCR Section 2.5.5
			• Trunk Group-Restore to Service (C)	• UCR Section 2.5.6

Table 3. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2-Wire Proprietary Digital	Yes No No	Access	• Directory Number Identification (R) • National ISDN 1/2 Basic Access (C) • Analog Line (R) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R) • Alerting Signals and Tones (R) • S/T Reference Point (ISDN BRI) (C)	• UCR Section 2.1.1 • UCR Section 2.3.3 • UCR Section 2.3.5 • UCR Section 2.5.4.1.1 • UCR Section 2.5.4.1.2 • UCR Section 5.2.1 • UCR Section 5.3.1 • UCR Section 5.5 • UCR Section 5.7.1.2.1
		Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (C) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don't Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (C)		• UCR Section 2.1 • UCR Section 2.1.3 • UCR Section 2.1.4 • UCR Section 2.1.5 • UCR Section 2.1.6 • UCR Section 2.1.7 • UCR Section 2.1.7.1 • UCR Section 2.1.7.2 • UCR Section 2.1.7.3 • UCR Section 2.1.7.4 • UCR Section 2.1.7.5 • UCR Section 2.1.7.6 • UCR Section 2.1.7.7 • UCR Section 2.1.7.8 • UCR Section 2.1.8.1 • UCR Section 2.1.8.2 • UCR Section 2.1.8.3 • UCR Section 2.1.8.4 • UCR Section 2.1.9 • UCR Section 2.7 • UCR Section 2.9
Attendant	No	• Attendant Features (C)		• UCR Section 2.2

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (C) • Outgoing call trace (C)	• UCR Section 2.4.1.1 • UCR Section 2.4.1.2 • UCR Section 2.4.1.3 • UCR Section 2.4.2 • UCR Section 2.4.3
Conferencing	Yes	• Preset Conferencing (C) • Meet-Me Conferencing (R) • Progressive Conferencing (C)	• UCR Section 2.6 • UCR Section 2.6.2 • UCR Section 2.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 2.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 2.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (C) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (C) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 3.1 • UCR Section 3.2 • UCR Section 3.2.1 • UCR Section 3.2.2 • UCR Section 3.3 • UCR Section 3.4.1 • UCR Section 3.4.2 • UCR Section 3.5 • UCR Section 3.6 • UCR Section 3.7 • UCR Section 3.8.1 • UCR Section 3.8.2 • UCR Section 3.8.3 • UCR Section 3.8.4 • UCR Section 3.8.5 • UCR Section 3.8.6 • UCR Section 3.8.7 • UCR Section 3.8.8 • UCR Section 3.8.9 • UCR Section 3.11

Table 3. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (C) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (C) • Digit Reception Requirements (R) • Screening (C)	• UCR Section 4.1 • UCR Section 4.2 • UCR Section 4.3.1 • UCR Section 4.3.2 • UCR Section 4.3.3 • UCR Section 4.3.4 • UCR Section 4.5.1.1 • UCR Section 4.5.1.2 • UCR Section 4.5.1.2.1 • UCR Section 4.5.1.2.2 • UCR Section 4.5.1.3 • UCR Section 4.5.1.3.1 • UCR Section 4.5.1.3.2 • UCR Section 4.5.1.3.3 • UCR Section 4.5.1.4 • UCR Section 4.5.1.5 • UCR Section 4.5.1.6 • UCR Section 4.5.1.7 • UCR Section 4.5.1.8.1 • UCR Section 4.5.1.8.2 • UCR Section 4.5.1.9 • UCR Section 4.5.2 • UCR Section 4.5.3 • UCR Section 4.5.4 • UCR Section 4.5.5 • UCR Section 4.5.6 • UCR Section 4.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (C) • Uniform Interface Configuration for BRIs (C) • Electronic Key Telephone Systems (EKTS) (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 10, Table 10-1 • UCR Section 10, Table 10-2 • UCR Section 10, Table 10-3 • UCR Section 10, Table 10-4 • UCR Section 10, Table 10-5 • UCR Section 10, Table 10-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 11.1.1.2 • UCR Section 11.1.2.2 • UCR Section 11.2 • UCR Section 11.3 • UCR Section 11.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 12.2 • UCR Section 12.3 • UCR Section 12.3.1 • UCR Section 12.3.2 • UCR Section 12.3.2.2 • UCR Section 12.3.3 • UCR Section 12.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 13

Table 3. PBX 1 Requirements (continued)

VoIP				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R)		• UCR App. 3, para. A3.2.1 • UCR App. 3, para. A3.2.2 • UCR App. 3, para. A3.2.3 • UCR App. 3, para. A3.2.4 • UCR App. 3, para. A3.2.5 • UCR App. 3, para. A3.2.6 • UCR App. 3, para. A3.2.7 • UCR App. 3, para. A3.2.8 • UCR App. 3, para. A3.2.9 • UCR App. 3, para. A3.2.10
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.2 • UCR Section 5.3.2 • UCR Section 5.3.4
NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.				

Table 3. PBX 1 Requirements (continued)

LEGEND:

ANSI	American National Standards Institute
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
CODEC	Coder/Decoder
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	Department of Defense Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0 (64 kbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E&M	Ear and Mouth
E1	European Basic Multiplex Rate (2.048 Mbps)
EKTS	Electronic Key Telephone System
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union- Telecommunication Standardization Sector
kbps	kilobits per second
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
min	minute
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
para.	paragraph
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PCM	Pulse Code Modulation
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.955.3	ISDN Signaling Standard for E1 MLPP
R	Required
S/T	ISDN BRI four-wire interface
SS7	Signaling System 7
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit -3rd generation
T.4	Standardization of Group 3 facsimile terminals for document transmission
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
UCR	Unified Capabilities Requirements
UPS	Uninterruptible Power Supply
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing
yr	year

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0814401.

FOR THE COMMANDER:

Steve Lasueski

2 Enclosures a/s

for RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Cisco Unified CallManager Version 4.3(2) Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7," 21 April 2009
(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007
(e) Defense Information Systems Agency, "Department of Defense Networks Unified Capabilities Requirements," 21 December 2007
(f) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Cisco Unified CallManager Version 4.3(2), Service Release (SR) 1b, with Internetwork Operating System (IOS) Software Release 12.4(15) T7; hereinafter referred to as the System Under Test (SUT).

2. PROPONENT. White House Communications Agency (WHCA).

3. PROGRAM MANAGER. Lt Col Lionel Ramos, WHCA/J5, 2743 Defense Blvd, Anacostia Annex, District of Columbia, 20373, e-mail: LLRamos@whmo.mil.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Private Branch Exchange (PBX) 1. The SUT supports American National Standards Institute (ANSI) T1.619a Digital Transmission Link Level 1 (T1) Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) National ISDN Standard 1 or 2 (NI 1/2), and International Telecommunication Union- Telecommunication Standardization Sector (ITU-T) Q.931 European Basic Multiplex Rate (E1) ISDN PRI interface. The SUT Voice over Internet Protocol (VoIP) configuration consists of CallManagers running the Cisco Unified CallManager software, gateways, and Internet Protocol (IP) telephones. The Cisco Unified CallManager is the software-based call-processing component of the Cisco enterprise IP telephone solution. The Cisco Unified CallManager software is a client-server application, loaded on a Personal Computer (PC) that is running a Cisco modified version of Windows 2003 Server. The Cisco CallManager software provides telephony features and capabilities to packet telephony network devices such as VoIP phones. The Cisco Unified CallManagers tested were the Media Convergence Server (MCS)7835H1, MCS7835H2, MCS7825H2, and MCS7825H3.

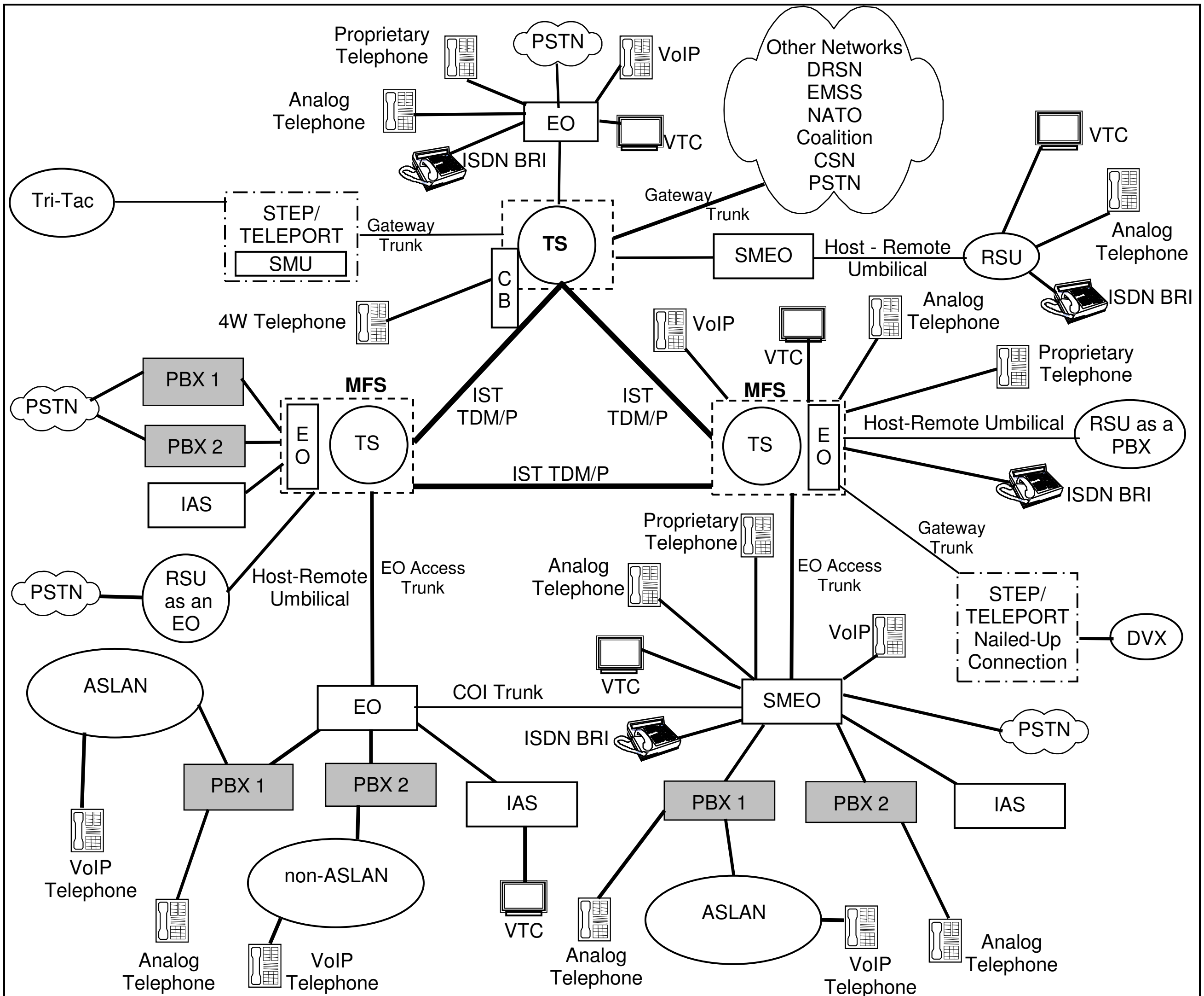
The 3745, 3845, and 2851 gateway routers are included in this tested architecture. The SUT gateway routers are scalable. The 2851 has one Network Module (NM) slot, one High-Density Extension Voice Module (EVM-HD) slot, and four High-Performance Wide Area Network (WAN) Interface Card (WIC) (HWIC) slots. These slots can be populated with up to 12 T1 trunks or 52 foreign-exchange-station (FXS) ports.

The 3845 has four NM slots and four HWIC slots. Each NM slot on the 3845 can accommodate a standard NM, an enhanced-network-module (NME) or an EVM-HD. The 3845 supports up to 24 T1 trunks or 88 FXS ports.

The 3745 has four NM slots and four WIC slots. The WIC slots are not capable of supporting Voice WICs (VWIC). The 3745 supports up to 18 T1 trunks or 48 FXS ports.

6. OPERATIONAL ARCHITECTURE. The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The

DSN architecture, therefore, consists of several categories of switches including PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.



LEGEND:

ASLAN	Assured Services Local Area Network	NATO	North Atlantic Treaty Organization
4W	4-Wire	PBX	Private Branch Exchange
BRI	Basic Rate Interface	PBX 1	Private Branch Exchange 1
CB	Channel Bank	PBX 2	Private Branch Exchange 2
COI	Community of Interest	PSTN	Public Switched Telephone Network
CSN	Canadian Switch Network	RSU	Remote Switching Unit
DRSN	Defense Red Switch Network	SMEO	Small End Office
DSN	Defense Switched Network	SMU	Switched Multiplex Unit
DVX	Deployable Voice Exchange	STEP	Standardized Tactical Entry Point
EMSS	Enhanced Mobile Satellite System	TDM/P	Time Division Multiplex/Packetized
EO	End Office	Tri-Tac	Tri-Service Tactical Communications Program
IAS	Integrated Access Switch	TS	Tandem Switch
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
IST	Interswitch Trunk	VTC	Video Teleconferencing
MFS	Multifunction Switch		System Under Test

Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 1s are listed in Table 2-1. These requirements are derived from:

a. DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)."

b. UCR interface and signaling requirements for trunks/lines verified through JITC testing and/or vendor submission of Letters of Compliance (LoC).

c. UCR PBX 1 Capability Requirements (CRs) and Feature Requirements (FRs) verified through JITC testing and/or vendor submission of LoC.

Table 2-1. PBX 1 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Trunking	• PBX Line (C) • Direct Inward Dialing (C) • National ISDN 1/2 Primary Access (R) • ISDN ANSI MLPP Service Capability (R) • ITU-T ISDN Primary Access (Europe only) (C) • ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (Europe only) (C) • Normal Wink Start Operations (C) • Glare Operation (C) • Abnormal Wink Start (C) • Glare Resolution (C) • Call for Service Timing (R) • Guard Timing (R) • Satellite Timing (C) • Disconnect Control (C) • Reselect and Retrial (C) • Off-Hook Supervision Transition (C) • Dial-Pulse Signals (C) • DTMF Signaling (C) • Standard Digit Format for Precedence (C) • MFR1 2/6 Signaling (C) • Alerting Signals and Tones (R) • DSN ISDN User-to-Network Signaling (R) • Application (R) • Physical Layer (R) • Data Link Layer (R) • Data Link Connection (R) • Peer-to-Peer Procedures of Data-Link Layer (R) • Layer 3 DSN User-to-Network Signaling (R) • DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R) • Sequence of Messages for DSN Circuit-Switched Calls (R) • Message Functional Definition and Content (R) • General Message Format and Information Elements Coding (R) • Supplementary Services (C) • PCM-24 Digital Trunk Interface (R) • Interface Characteristics (R) • Supervisory Channel Associated Signaling (C) • Clear Channel Capability (R) • Alarm and Restoral Requirements (R) • PCM-30 Digital Trunk Interface (Europe only) (C) • Interoperation of PCM-24 and PCM-30 (C) • Analog Trunk Interface (C) • Integrated Digital Loop Carrier (C) • Trunk Group-Remove from Service (C) • Trunk Group-Restore to Service (C)	• UCR Section 2.3.1 • UCR Section 2.3.2 • UCR Section 2.3.4.1 • UCR Section 2.3.4.1.1 • UCR Section 2.3.4.2 • UCR Section 2.3.4.2.1 • UCR Section 5.3.3.1.1 • UCR Section 5.3.3.1.2 • UCR Section 5.3.3.2.1 • UCR Section 5.3.3.2.2 • UCR Section 5.3.5 • UCR Section 5.3.6 • UCR Section 5.3.7 • UCR Section 5.3.8 • UCR Section 5.3.9 • UCR Section 5.3.10 • UCR Section 5.4.1 • UCR Section 5.4.2 • UCR Section 5.4.2.1 • UCR Section 5.4.3 • UCR Section 5.5 • UCR Section 5.7.1 • UCR Section 5.7.1.1 • UCR Section 5.7.1.2 • UCR Section 5.7.1.3 • UCR Section 5.7.1.3.1 • UCR Section 5.7.1.3.2 • UCR Section 5.7.1.4 • UCR Section 5.7.1.4.2 • UCR Section 5.7.1.4.3 • UCR Section 5.7.1.4.4 • UCR Section 5.7.1.4.5 • UCR Section 5.7.1.4.6 • UCR Section 7.1 • UCR Section 7.1.1 • UCR Section 7.1.2 • UCR Section 7.1.3 • UCR Section 7.1.4 • UCR Section 7.2 • UCR Section 7.3 • UCR Section 7.4 • UCR Section 7.5 • UCR Section 2.5.5 • UCR Section 2.5.6

Table 2-1. PBX 1 Requirements (continued)

DSN Trunk Interfaces (continued)				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.619a) E1 ISDN PRI (ITU-T Q.955.3)	No No (Europe only) Yes No (Europe only)	Voice	• MOS (R) • Secure calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • 56 kbps switched data (R: PRI only) • 64 kbps switched data (R: PRI only) • NX56 synchronous BER (R: PRI only) • NX64 synchronous BER (R: PRI only) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • UCR Section 3.10 • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: PRI only)	• FTR 1080B-2002
DSN Line Interfaces				
2-Wire Analog ISDN BRI NI 1/2 (ANSI T1.619a) 2-Wire Proprietary Digital	Yes No No	Access	• Directory Number Identification (R) • National ISDN 1/2 Basic Access (C) • Analog Line (R) • Basic Line Test Capabilities (R) • Advanced Line Test Capabilities (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (R) • Alerting Signals and Tones (R) • S/T Reference Point (ISDN BRI) (C)	• UCR Section 2.1.1 • UCR Section 2.3.3 • UCR Section 2.3.5 • UCR Section 2.5.4.1.1 • UCR Section 2.5.4.1.2 • UCR Section 5.2.1 • UCR Section 5.3.1 • UCR Section 5.5 • UCR Section 5.7.1.2.1
		Voice	• MOS (R) • Secure Calls (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • Secure data (STE/STU-III) (R)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (R: BRI only)	• FTR 1080B-2002

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Individual Lines (R) • Denied originating service (C) • Code restriction and diversion (C) • Call waiting (R) • Three-way calling (R) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (R) • Call Transfer - Internal Only (R) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R) • Call Transfer – Outside (R) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (R) • Conference Calling – Six Way Station Controlled (C) • Call forwarding Variable (R) • Call Forward Busy Line (R) • Call Forwarding – Don’t Answer – All Calls (R) • Selective Call Forwarding (C) • Call pick-up (C) • Address Translation (C) • Assured Dial Tone (C)	• UCR Section 2.1 • UCR Section 2.1.3 • UCR Section 2.1.4 • UCR Section 2.1.5 • UCR Section 2.1.6 • UCR Section 2.1.7 • UCR Section 2.1.7.1 • UCR Section 2.1.7.2 • UCR Section 2.1.7.3 • UCR Section 2.1.7.4 • UCR Section 2.1.7.5 • UCR Section 2.1.7.6 • UCR Section 2.1.7.7 • UCR Section 2.1.7.8 • UCR Section 2.1.8.1 • UCR Section 2.1.8.2 • UCR Section 2.1.8.3 • UCR Section 2.1.8.4 • UCR Section 2.1.9 • UCR Section 2.7 • UCR Section 2.9
Attendant	No	• Attendant Features (C)	• UCR Section 2.2

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Service (C) • Enhanced Emergency Service (E911) (C) • Trace of terminating calls (C) • Outgoing call trace (C)	• UCR Section 2.4.1.1 • UCR Section 2.4.1.2 • UCR Section 2.4.1.3 • UCR Section 2.4.2 • UCR Section 2.4.3
Conferencing	Yes	• Preset Conferencing (C) • Meet-Me Conferencing (R) • Progressive Conferencing (C)	• UCR Section 2.6 • UCR Section 2.6.2 • UCR Section 2.6.3
Nailed-up Connections	No	• Nailed-Up Connections (C)	• UCR Section 2.8
DSN Hotline Services	No	• DSN Analog Hotline Service (C)	• UCR Section 2.12
MLPP	Yes	• MLPP Overview (R) • Preemption in the Network (R) • Network Facility with Lower Precedence Calls (R) • Network Facility with Equal or Higher Precedence Calls (R) • Precedence Call Diversion (R) • Channel Associated Signaling (C) • Primary Rate Interface (R) • Analog Line MLPP (R) • ISDN MLPP Basic Rate Interface (C) • ISDN Primary Rate Interface (R) • Precedence Call Waiting (R) • Call Forwarding (R) • Call Transfer (R) • Call Hold (R) • Three-Way Calling (R) • Call Pickup (C) • Conferencing (C) • Multiline Hunt Group (C) • Community of Interest (C) • MLPP Interaction with EKTS features (C)	• UCR Section 3.1 • UCR Section 3.2 • UCR Section 3.2.1 • UCR Section 3.2.2 • UCR Section 3.3 • UCR Section 3.4.1 • UCR Section 3.4.2 • UCR Section 3.5 • UCR Section 3.6 • UCR Section 3.7 • UCR Section 3.8.1 • UCR Section 3.8.2 • UCR Section 3.8.3 • UCR Section 3.8.4 • UCR Section 3.8.5 • UCR Section 3.8.6 • UCR Section 3.8.7 • UCR Section 3.8.8 • UCR Section 3.8.9 • UCR Section 3.11

Table 2-1. PBX 1 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Call Processing	Yes	• Call Treatments (R) • Primary and Alternate Routing (C) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (R) • Termination of Analog Lines (R) • DSN User Dialing (R) • Interswitch and Intraswitch Dialing (R) • Seven-Digit Dialing (R) • Ten-Digit Dialing (R) • Access Code (R) • Access Digit (R) • Precedence Digit (R) • Service Digit (R) • Route Code (R) • Area Code (R) • Switch Code (R) • Line Number (R) • Calling Name Delivery (C) • Calling Number Delivery (R) • Emergency Service 911 Conflict Resolution (R) • DSN Switch Outpulsing Digit Formats (C) • Standard Directory Number (R) • Standard Test Numbers (C) • Base Services – Abbreviated Numbers (C) • Digit Reception Requirements (R) • Screening (C)	• UCR Section 4.1 • UCR Section 4.2 • UCR Section 4.3.1 • UCR Section 4.3.2 • UCR Section 4.3.3 • UCR Section 4.3.4 • UCR Section 4.5.1.1 • UCR Section 4.5.1.2 • UCR Section 4.5.1.2.1 • UCR Section 4.5.1.2.2 • UCR Section 4.5.1.3 • UCR Section 4.5.1.3.1 • UCR Section 4.5.1.3.2 • UCR Section 4.5.1.3.3 • UCR Section 4.5.1.4 • UCR Section 4.5.1.5 • UCR Section 4.5.1.6 • UCR Section 4.5.1.7 • UCR Section 4.5.1.8.1 • UCR Section 4.5.1.8.2 • UCR Section 4.5.1.9 • UCR Section 4.5.2 • UCR Section 4.5.3 • UCR Section 4.5.4 • UCR Section 4.5.5 • UCR Section 4.5.6 • UCR Section 4.5.8
ISDN Services	Yes	• BRI Access, Call Control and Signaling (C) • Uniform Interface Configuration for BRIs (C) • Electronic Key Telephone Systems (EKTS) (C) • PRI Access, Call Control and Signaling (R) • PRI Features (R) • Packet Data Features and Capabilities (C)	• UCR Section 10, Table 10-1 • UCR Section 10, Table 10-2 • UCR Section 10, Table 10-3 • UCR Section 10, Table 10-4 • UCR Section 10, Table 10-5 • UCR Section 10, Table 10-6
Synchronization	Yes	• Line timing mode (R) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 11.1.1.2 • UCR Section 11.1.2.2 • UCR Section 11.2 • UCR Section 11.3 • UCR Section 11.4
Reliability	Yes	• System Availability (R) • Backup Power (R) • Power Components (R) • UPS Requirements (R) • UPS PBX 1 Load Capacity (R) • Backup Power (Environmental) (R) • Alarms (R)	• UCR Section 12.2 • UCR Section 12.3 • UCR Section 12.3.1 • UCR Section 12.3.2 • UCR Section 12.3.2.2 • UCR Section 12.3.3 • UCR Section 12.3.4
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Section 13

Table 2-1. PBX 1 Requirements (continued)

VoIP			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
VoIP System	No	VoIP function is conditional. If VoIP is provided, **all** of the following requirements must be met: • Voice Quality with MOS of 4.0 or better (R) • ITU-T G.711 PCM CODEC (R) • MLPP (R) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging (R) • VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr) (R)	• UCR App. 3, para. A3.2.1 • UCR App. 3, para. A3.2.2 • UCR App. 3, para. A3.2.3 • UCR App. 3, para. A3.2.4 • UCR App. 3, para. A3.2.5 • UCR App. 3, para. A3.2.6 • UCR App. 3, para. A3.2.7 • UCR App. 3, para. A3.2.8 • UCR App. 3, para. A3.2.9 • UCR App. 3, para. A3.2.10

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN (See note.)	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (R) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.2 • UCR Section 5.3.2 • UCR Section 5.3.4

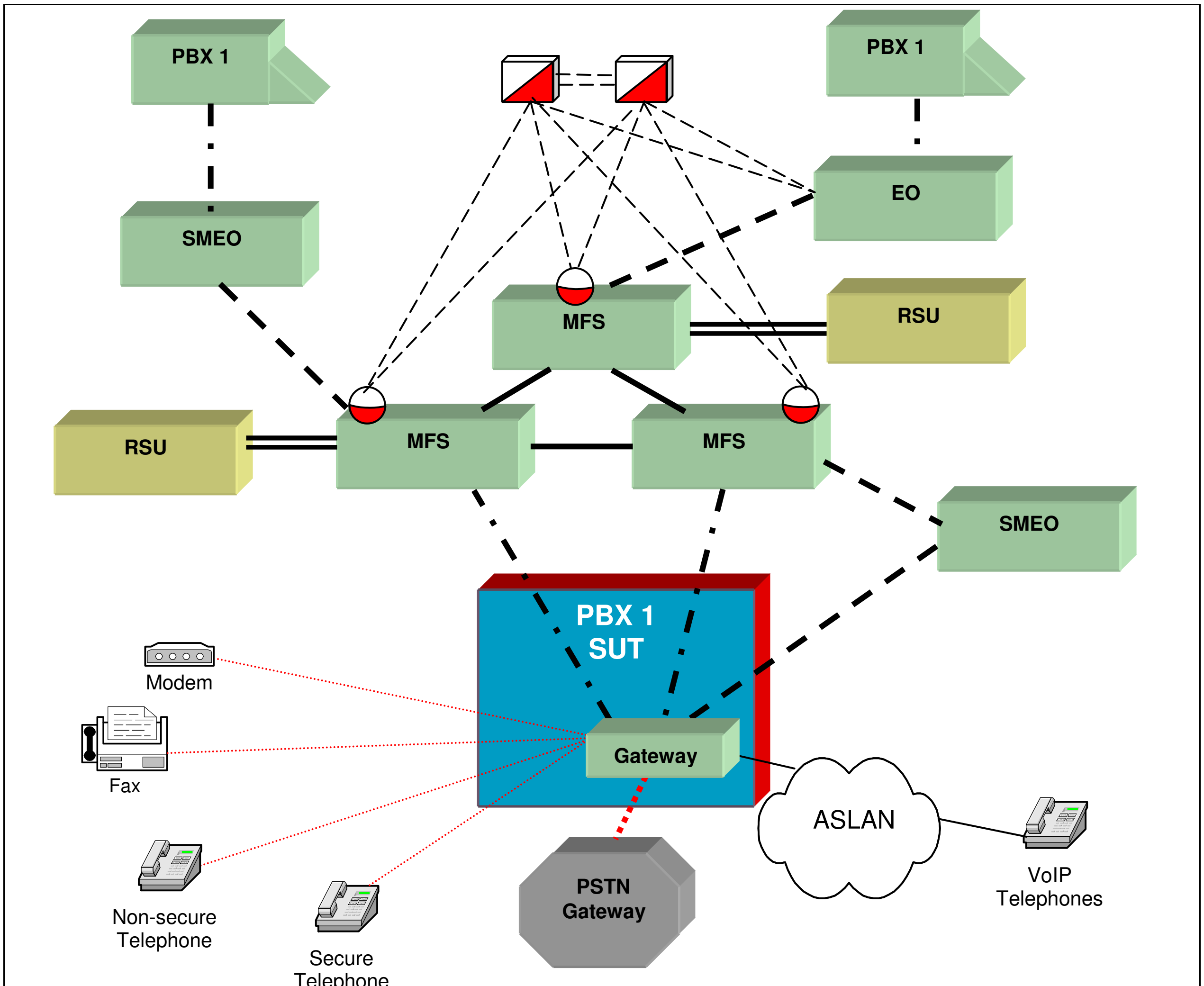
NOTE: Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

Table 2-1. PBX 1 Requirements (continued)

LEGEND:

ANSI	American National Standards Institute
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
CODEC	Coder/Decoder
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	Department of Defense Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0 (64 kbps)
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSN	Defense Switched Network
DTMF	Dual Tone Multi-Frequency
E&M	Ear and Mouth
E1	European Basic Multiplex Rate (2.048 Mbps)
EKTS	Electronic Key Telephone System
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union-Telecommunication Standardization Sector
kbps	kilobits per second
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
min	minute
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
para.	paragraph
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange 1
PCM	Pulse Code Modulation
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.955.3	ISDN Signaling Standard for E1 MLPP
R	Required
S/T	ISDN BRI four-wire interface
SS7	Signaling System 7
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit - 3rd generation
T.4	Standardization of Group 3 facsimile terminals for document transmission
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
UCR	Unified Capabilities Requirements
UPS	Uninterruptible Power Supply
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing
yr	year

8.   TEST NETWORK DESCRIPTION. The SUT was tested at JITC's Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system's required functions and features was conducted using the test configuration depicted in Figure 2-2. The PBX 1 test configuration with an Assured Services Local Area Network (ASLAN) is depicted in Figure 2-3. The SUT was tested as the end-point in relation to the other switches.



LEGEND:

ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
CAS	Channel Associated Signaling
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EO	End Office
FAX	facsimile
ISDN	Integrated Services Digital Network
Mbps	Megabits per second
MFS	Multifunction Switch
PBX	Private Branch Exchange
PBX 1	Private Branch Exchange Type 1
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RSU	Remote Switching Unit
SMEO	Small End Office
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TCP/IP	Transmission Control Protocol/Internet Protocol
VoIP	Voice over Internet Protocol

SS7 Service Switching Point (SSP)

SS7 Signal Transfer Point (STP)

PSTN Gateway Trunk

DSN Interswitch Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

DSN Line (2 Wire Analog, ISDN BRI, Digital Proprietary)

SS7 Links (A-Link, B-Link, or C-Link)

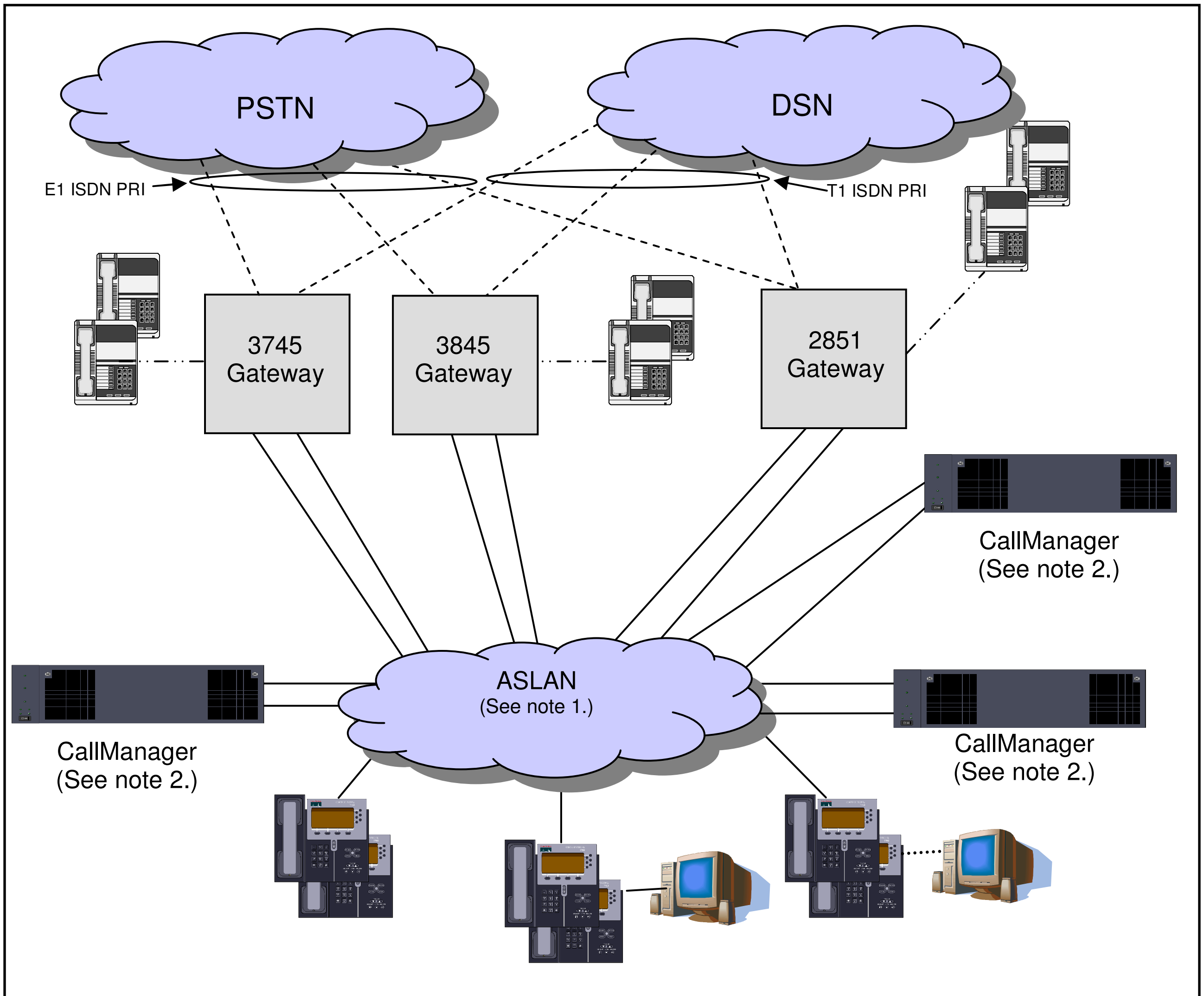
TCP/IP

DSN End Office Access Trunk (T1/E1 SS7, T1/E1 CAS, T1/E1 ISDN PRI)

DSN PBX Access Trunk (T1/E1 CAS, T1/E1 ISDN PRI)

RSU-Host Umbilical Link

Figure 2-2. Test Configuration



LEGEND:

ASLAN	Assured Services Local Area Network
CP	Cisco Phone
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
ISDN	Integrated Services Digital Network
Mbps	Megabits per second
MCS	Media Convergence Server
PBX 1	Private Branch Exchange 1
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
VoIP	Voice over Internet Protocol

4-Wire Digital TDM Interfaces
100 Mbps Ethernet
2-Wire Analog
1 Gigabit Ethernet
CP79xx VoIP Telephones (See note 3.)
Analog Telephones
PC for shared access

NOTES:
1 The SUT is certified with any ASLAN listed on the Unified Capabilities Approved Products List.
2 The following CallManagers were tested: MCS7835H1, MCS7835H2, MCS7825H2, and MCS7825H3.
3 Refer to paragraph 11.a.(5)(a)10.c. for certified shared access rates.

Figure 2-3. PBX 1 with ASLAN

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

Table 2-2. Tested System Configurations

<table>
<tr><th colspan="2">System Name</th><th colspan="2">Software Release</th></tr>
<tr><td colspan="2">Nortel CS2100</td><td colspan="2">Succession Enterprise (SE) 09.1</td></tr>
<tr><td colspan="2">Siemens EWSD</td><td colspan="2">19d with Patch Set 46</td></tr>
<tr><td colspan="2">Alcatel-Lucent 5ESS</td><td colspan="2">5E16.2 Broadcast Warning Message (BWM) 07-0003</td></tr>
<tr><td colspan="2">Avaya S8710</td><td colspan="2">Communication Manager (CM) 4.0 (R014x.00.2.731.7: Super Patch 14419)</td></tr>
<tr><td colspan="2">CUCCX</td><td colspan="2">6.0</td></tr>
<tr><td colspan="2">Cisco IP Communicator</td><td colspan="2">2.1.4.5</td></tr>
<tr><th colspan="4">Cisco Unified CallManager Version 4.3(2) SR1b, with IOS Software Release 12.4(15) T7</th></tr>
<tr><th>Component
(See note 1.)</th><th>Release</th><th>Sub-component
(See note 1.)</th><th>Function</th></tr>
<tr><td>CallManagers
MCS7835H2, MCS7825H3, MCS7825H2, MCS7835H1, MCS7835H, MCS7835I1, MCS7845H2, MCS7825H, MCS7835I, MCS7845H, MCS7845I, MCS7825-H1, MCS7825I1, MCS7845H1, MCS7845I1</td><td>4.3(2) SR1b</td><td>Not Applicable</td><td>Processing/Signaling</td></tr>
<tr><td rowspan="7">Cisco 3745/3725 Multiservice Access Router (Gateway)
(See note 2.)</td><td rowspan="7">IOS 12.4(15) T7</td><td>NM HD 2V</td><td>TDM Interface NM HD Voice, 2-slot IP communications voice/fax</td></tr>
<tr><td>NM HD 2VE</td><td>TDM Interface NM HD Voice, 2-slot IP communications enhanced voice/fax</td></tr>
<tr><td>VIC 4FXS/DID</td><td>Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, Direct Inward Dial</td></tr>
<tr><td>VIC2 2FXS</td><td>Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station</td></tr>
<tr><td>VWIC 1MFT T1</td><td>Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1</td></tr>
<tr><td>VWIC 2MFT T1</td><td>Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1</td></tr>
<tr><td>VWIC 2MFT T1 DI</td><td>Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1, Drop and Insert</td></tr>
</table>

Table 2-2. Tested System Configurations (continued)

Cisco Unified CallManager Version 4.3(2) SR1b, with IOS Software Release 12.4(15) T7			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
Cisco 3845/3825 Integrated Services Router (Gateway)	IOS 12.4(15) T7	**NM HDV2**	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/ E1 (See note 3.)
Cisco 2851 Integrated Services Router (Gateway)	IOS 12.4(15) T7	NM HDV2	TDM Interface NM, HD Voice, 2-slot IP communications enhanced voice/fax
		VIC 4FXS/DID	Voice Interface Card, 4-port, RJ-11, Foreign Exchange Station, DID
		VWIC2 2MFT T1/E1	Second Generation Voice/WAN Interface Card 2-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		EVM HD 8FXS/DID	HD analog and digital extension module for voice and fax
		EM HDA 8FXS	8-port analog Foreign Exchange Station expansion module for voice and fax (See note 4.)
		NM HDV2 2T1/E1	2-port T1/E1 IP Communications HD voice/fax NM, 2 T1/E1 controllers (See note 3.)
		NM HDV2 1T1/E1	1-port T1/E1 IP Communications HD voice/fax NM, 1 T1/E1 controllers (See note 3.)
		VWIC2 1MFT T1/E1	Second Generation Voice/WAN Interface Card 1-port RJ-48, Multiflex Trunk T1/E1 (See note 3.)
		VIC2 2FXS	Voice Interface Card, 2-port, RJ-11, Foreign Exchange Station
CP-7940G and CP-7960G	Load: P00308000900	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7970G and CP-7971G	Load: SCCP70.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7911G and CP-7906G	Load: SCCP11.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7941G, CP-7941G-GE, CP-7961G, and CP-7961G-GE	Load: SCCP41.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7942G and CP-7962G	Load: SCCP42.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7945G and CP-7965G	Load: SCCP45.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
CP-7975G	Load: SCCP75.8-4-1SR1S	Not Applicable	IP Phone (with push-to-talk handset or with standard handset)
7914	Load: S00105000300	Not Applicable	Expansion module
7915	B015-1-0-2	Not Applicable	Expansion module

Table 2-2. Tested System Configurations (continued)

Cisco Unified CallManager Version 4.3(2) SR1b, with IOS Software Release 12.4(15) T7			
Component (See note 1.)	**Release**	**Sub-component (See note 1.)**	**Function**
7916	B016-1-0-2SR1	Not Applicable	Expansion module
CIS 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	CP-7961G IP phone, TEMPEST version
CIS 7975G (See note 5.)	SCCP75.8-4-1SR1S	Not Applicable	7975G IP phone, TEMPEST version
CRYPTEK 7961G (See note 5.)	SCCP41.8-4-1SR1S	Not Applicable	7961G IP phone, TEMPEST version with no PC interface and no shared access
Walker WS-2620	Not Applicable	Not Applicable	Push to Talk Handset for Cisco 7900 Series phones

NOTES:

1 Components bolded and underlined were tested by JITC. The other components in the family series were not tested; however, they utilize the same IOS software and hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes and they are also certified for joint use.

2 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.

b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.

c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP Phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

3 These components are certified in the DSN with T1 ISDN PRI interface. These components are certified in the PSTN with the T1 ISDN PRI and E1 ISDN PRI interfaces.

4 The EM HDA 8FXS expansion module requires the EVM HD module. Up to two EM HDA 8FXS expansion modules are supported for each EVM HD.

5 CIS Secure Computing and Cryptek add security modifications which are physical in nature and do not affect software. The CIS Secure Computing and Cryptek security modifications can be applied to any certified Cisco IP phone.

LEGEND:

5ESS	Class 5 Electronic Switching System
10/100BaseT	10/100 Mbps (Baseband Operation, Twisted Pair) Ethernet
CP	Cisco Phone
CS	Communication Server
CUCCX	Cisco Unified Contact Center Express
DI	Drop and Insert
DID	Direct Inward Dialing
DSN	Defense Switched Network
E1	European Basic Multiplex Rate (2.048 Mbps)
EM	Expansion Module
EVM	Extension Voice Module
Fax	facsimile
FXS	Foreign Exchange Station
EWSD	Elektronisches Wählsystem Digital
G	10/100BaseT Ethernet (A Cisco part designator on their IP phone.)
GE	Gigabit Ethernet (A Cisco part designator on their IP phone.)
HD	High Density
HDA	High Density Analog
IOS	Internetwork Operating System
IP	Internet Protocol
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
JITC	Joint Interoperability Test Command
Mbps	Megabits per second
MCS	Media Convergence Server
MFT	Multiflex Trunk
ms	milliseconds
NM	Network Module
PBX 1	Private Branch Exchange 1
PC	Personal Computer
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
RJ	Registered Jack
SCCP	Skinny Client Control Protocol
SR	Service Release
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
TDM	Time Division Multiplexing
UCR	Unified Capabilities Requirements
V	Voice
VE	Voice/Fax Enhanced
VIC	Voice Interface Card
VWIC	Voice WAN Interface Card
WAN	Wide Area Network

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces

(a) The SUT met all critical CRs and FRs for T1 ISDN PRI NI 1/2 ANSI T1.619a interface with the following minor exceptions:

1. ROUTINE calls that are placed over a trunk that is broken do not receive an Isolated Code Announcement (ICA), but instead receive a T-120 busy. Calls above ROUTINE attempted over a trunk that is broken do not receive an ICA. The operational impact is minor because they are treated with a Blocked Precedence Announcement (BPA) and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

2. The SUT does not support Non Facility Associated Signaling (NFAS) on their ISDN PRI National ISDN Standard 2 (NI2) interface. The Defense Information Systems Agency's (DISA's) adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

3. A discrepancy exists that is associated with the monitoring tool that SUT uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. However, due to the fact that the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, the operational impact is minor.

(b) T1 CAS is supported by the SUT; however, it was not tested with this software release. Critical interoperability discrepancies were discovered during testing of a previous software release that have not been fixed by the vendor. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN Program Management Office (PMO) for use within the DSN. This is not a required interface for a PBX 1. The SUT did not meet the critical requirement listed below:

1. T1 CAS wink start recognition is not within the required tolerance of 100 milliseconds (ms) to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms.

2. The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid.

3. During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator.

4. Calls that are attempted over a trunk that is broken or in a remote busy-out condition receive a BPA received in lieu of an ICA.

5. When channels on a T1 CAS trunk group are busied by the remote switching system, the SUT fails to acknowledge all of these busy outs. As a result calls originated by the SUT fail to complete and the proper treatment is not provided.

(c) E1 ISDN PRI is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. Therefore, this interface is not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

(2) DSN Line Interfaces. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE) and VoIP DSN line interfaces. The following paragraphs detail the discrepancies that have a minor operational impact for interoperability that were discovered during testing:

(a) The SUT does not support a Multi-Level Precedence and Preemption (MLPP) global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified CallManager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

(b) When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the Busy Not Equipped Announcement (BNEA) is not provided unless the attendant or alternate destination is busy. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.

(c) To meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal, a configuration change was required on the analog gateways. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

(d) When an analog station is active with a call and is preempted by a higher precedence call, the analog station receives the proper Preemption Notification Tone (PNT). However, after going on hook, the station rings at ROUTINE. This was determined to be a minor impact because the station is still preempted correctly.

(3) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Features and Capabilities with the following exceptions: Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX 1 for all instruments; however, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore are not covered in this certification. This feature is not required for a PBX 1. There is no risk associated with the SUT not supporting these features. Refer to Table 2-3 for a list of the Common Features supported for the phone types and associated testing observations.

Table 2-3. SUT Common Call Feature Availability

Call Feature	Phone Type	
	Analog	IP[1]
Precedence Call Waiting	Not Supported[2]	Not Supported[3]
Call Hold	Not Supported[2]	Passed
Call Forwarding No Answer	Passed	Passed
Call Forwarding Busy	Passed	Passed
Call Forwarding Variable	Not Supported[2]	Passed[4]
Three-Way Calling	Not Supported[2]	Passed[5]
Call Transfer	Not Supported[2]	Passed
Multi-line Hunt Service	Passed[5]	Passed[6]
Call Pickup	Not Supported	Passed
Denied Originating Service	Not Supported[7]	Not Supported[7]

NOTES:

1 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions. There is no operational impact.

2 The SUT analog gateway does not support the following required line features: Call Transfer, Call Hold, Precedence Call Waiting, Call Forwarding Variable, Three-Way Calling, and Call Pickup. This is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

3 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

4 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

5 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. DISA adjudicated this anomaly as having a minor operational impact because the preempted user receives PNT and the other members remain connected.

6 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. There is no operational impact.

7 Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a PBX 1.

LEGEND:

DISA Defense Information Systems Agency
IP Internet Protocol
MLPP Multi-Level Precedence and Preemption
PBX 1 Private Branch Exchange 1
PNT Precedence Notification Tone
SUT System Under Test
UCR Unified Capabilities Requirements
VoIP Voice over Internet Protocol

(b) Attendant. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(c) Public Safety. The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

(d) Conferencing. Meet-Me Conferencing is met through the use of an optional adjunct conferencing system call the Cisco MeetingPlace which is covered under a separate certification. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.

(e) Nailed-up Connections. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(f) DSN Hotline Services. This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

(g) MLPP. Met all critical CRs and FRs with the following minor exception: The SUT does not support the Loss of Command and Control (C2) announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

(h) Call Processing. Met all critical CRs and FRs.

(i) ISDN Services. Met all critical CRs and FRs. The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

(j) Synchronization. All critical interoperability certification CRs and FRs were met for this feature by the SUT. The SUT supports line timing mode and Internal Stratum 4 for synchronization.

(k) Reliability. All critical interoperability certification CRs and FRs for this feature were met by the SUT and met by vendor LoC.

(l) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

(4) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateways. The interfaces certified for the PSTN are T1 ISDN PRI NI 1/2 (ANSI T1.607), ITU-T Q.931 E1 ISDN PRI, and 2-Wire Analog Ground Start Line (GR-506 CORE). The SUT offers a T1 CAS trunk interface; however, it was not tested with this software release. Critical interoperability discrepancies were discovered during testing of a previous software release. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1. Refer to paragraph 11.a.(1)(b) for specific information.

(5) VoIP. The SUT is certified with any ASLAN on the UC APL.

(a) VoIP System. The UCR, appendix 3, section A3.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR, appendix 3, section A3.2.1, VoIP calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with ITU-T P.800 voice quality standards. This applies from handset to handset and from handset to gateway trunk in the DSN. For intra-switch calls, the SUT VoIP solution had an average MOS of 4.34 with a minimum measured MOS value of 4.13. The average inter-switch MOS was 4.35 with a minimum measured MOS value of 4.06. This average was based on a total of 930 calls. Additionally, VoIP systems shall not lose more than 150 ms of voice media in any five-minute period. This applies from handset to handset and from handset to gateway trunk to the DSN. The SUT met this requirement with a loss of no more than 0.00 ms of voice media packets in any five-minute period.

2. Codec. In accordance with the UCR, appendix 3, section A3.2.2, the International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill was required and was met by the SUT VoIP solution.

3. Multi-Level Precedence and Preemption (MLPP). In accordance with the UCR, appendix 3, section A3.2.3, the VoIP system shall meet all MLPP requirements identified in UCR, section 3. All critical MLPP features and functions were met by the SUT.

4. Security. Security requirements in accordance with the UCR, appendix 3, section A3.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, reference (c).

5. Network Management (NM). In accordance with the UCR, appendix 3, section A3.2.5, the vendor is required to provide a management system to monitor the performance of the ASLAN portion of the VoIP system. This requirement was verified via a LoC because of the numerous third party systems and applications capable of performing this function. The switching system NM requirements in accordance with the UCR, section 9, are not required for a PBX 1 and were not tested.

6. Synchronization. In accordance with the UCR, appendix 3, section A3.2.6, the VoIP system shall meet all synchronization requirements identified in UCR, section 11. The SUT derived synchronization with line timing mode via traditional T1 Time Division Multiplexing (TDM)-based interfaces.

7. Latency. The UCR, appendix 3, section A3.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP handset to the egress trunk. The SUT average latency over 930 inter-switch calls, with a minimum duration of 5 minutes for each call, was measured to be 54.63 ms.

8. Internet Protocol version 6 (IPv6). An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of Internet Protocol version 4 (IPv4). IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of their respective company. The vendor stated, in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).

b. Maintaining interoperability in heterogeneous environments and with IPv4.

c. Commitment to upgrade as the IPv6 standard evolves.

d. Availability of contractor/vendor IPv6 technical support.

All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP-7960G do not meet the critical IPv6 capability requirement in accordance with the GSCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

- The component must already be JITC certified and currently fielded within the DSN.
- This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.
- There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP Phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

9. In accordance with the UCR, appendix 3, section A3.2.9.1, the VoIP system components shall meet the following requirements:

a. All components shall be capable of implementing Service Class tagging using the 6-bit Differentiated Services Code Points (DSCPs) field in the IP header. The SUT end instruments used 6-bit service class tagging in the IP header, which meets the requirement.

b. All components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Tables A3-1 and A3-2. The VoIP SUT solution has the ability to assign any DSCD value from 0-63, which meets the requirement.

c. Any component that supports Real Time traffic and data shall be capable of tagging all Real Time traffic with an Institute of Electrical and Electronics Engineers (IEEE) 802.1Q 2-byte Tag Control Information (TGI) field 12-bit virtual LAN (VLAN) Identification (VID). The VoIP SUT solution supports Real Time traffic. Data was not mixed with Real Time traffic, so tagging was conditional.

10. In accordance with the UCR, appendix 3, section A3.2.9.2, the VoIP system end user devices shall meet the following requirements:

a. All end instrument components shall be capable of implementing Service Class tagging using the 6-bit DSCPs field in the IP header. The SUT end instruments used 6-bit service class tagging in the IP header, which meets the requirement.

b. The DSCPs shall be assigned to any distinct service class that originates or traverses the end instrument. The DSCPs may be assigned by either having the end instrument itself assign the DSCP to the distinct service class or having the call control portion of the VoIP system tell the end instrument what DSCP to insert to the distinct service class. The SUT end instrument assigned a DSCP value of 48 for voice signaling and 46 for voice media, which meets the requirement.

c. Any end instrument that supports Real Time traffic shall be capable of tagging all Real Time traffic with an IEEE 802.1Q 2-byte TCI field 12-bit VID. The SUT tagged the voice VID with 114 and the data VID with 11, which meets the requirement. The Cisco VoIP phones that met the critical interoperability requirements for certification were the CP7906G, CP7911G, CP7940G, CP7941G, CP7941G-GE, CP7942G, CP7945G, CP7960G, CP7961G-GE, CP7961G, CP7962G, CP7965G, CP7970G, CP7971G-GE, CP7975G, Tempest phone Cryptek 7961G, Tempest phone CIS 7961G, and Tempest phone CIS 7975G. The above phones have been tested and are certified for 100 Mbps shared access (i.e., same switch port is shared by PC and IP phone) with the exception of the CP7906G. The CP7906G phone does not support shared access. The following phones were tested and are certified for 1 gig shared access: CP7971G-GE, CP7975G, CP7965G, CP7945G, CP7941G-GE, CP7961G-GE, and Tempest phone CIS 7975G. The Tempest phones Cryptek 7961G, and CIS 7961G must have "Port Policing" configured at the network interface in order to allow proper port shared access. The CP7970G and CP7971G-GE phones are capable of web browsing; however, this feature was not tested, is not covered by this certification, and is not authorized for use within the DSN. All VoIP phones were tested using Secure Real Time Protocol (SRTP) which encrypts the media stream. The SRTP is able to encrypt only IP phone to IP phone intra-switch traffic and IP phone to gateway intra-switch traffic. All other calls (i.e. analog to analog, or analog to gateway traffic) are not encrypted.

11. In accordance with the UCR, appendix 3, section A3.2.10, the VoIP system shall meet the maximum downtime of 80 minutes per year for the system and 120 minutes per year for the subscriber. This requirement was met via a LoC.

(b) Scalability. The MCS7835s can support a maximum of 2,500 IP subscribers, the MCS7825 can support 1,000 IP subscribers, and the MCS7845 can support 7,500 IP subscribers. However; the configurations range from 1,000 IP subscribers with two MCS7825s to 30,000 subscribers with eight MCS7845s. The recommendation is to consult an engineer to determine the appropriate configurations. The SUT is certified with any certified ASLAN on the UC APL. The ASLAN can be scaled to meet the maximum subscribers as long as it is comprised of the equipment and software listed in this certification, and meets the traffic engineering constraints contained in the UCR, appendix 3.

b. System Interoperability Results. The SUT is certified for joint use in the DSN as a PBX 1 and PBX 2 in accordance with the requirements set forth in the UCR. The identified test discrepancies that remained open after software patches were applied and regression testing was completed have an overall minor operational impact. The following CallManagers were not tested; however, they utilize the same IOS software and hardware: MCS7825H, MCS7825H1, MCS7825I1, MCS7835H, MCS7835I, MCS7835I1, MCS7845H, MCS7845H1, MCS7845H2, MCS7845I, and MCS7845I1. JITC analysis determined them to be functionally identical for interoperability certification purposes. The 3725 and 3825 gateway routers were not tested; however, they utilize the same IOS software and hardware. JITC analysis determined them to be functionally identical for interoperability certification purposes. The SUT interoperability test summary is shown in Table 2-4. The SUT Interoperability Requirements/Status is shown in Table 2-5.

Table 2-4. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release and have not been fixed by the vendor. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA.[4] When channels on a T1 CAS trunk group are busied by the remote switching system, the SUT fails to acknowledge all of these busy outs. As a result calls originated by the SUT fail to complete and the proper treatment is not provided. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: Calls above ROUTINE that are attempted over a trunk that is broken receive a BPA in lieu of an ICA.[4] This interface does not support NFAS.[5] The SUT monitoring tool occasionally provides inaccurate reports when a remote trunk is busy.[6]
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not Tested	The E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog Loop Start (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with a minor configuration change[7] and the following minor exceptions: The SUT does not support an MLPP global diversion number.[8] Diversion to the alternate directory number is provided in lieu of the BNEA.[9] When an analog line is preempted at a precedence higher than the already established call, the analog interface will ring at ROUTINE.[10] The operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.
VoIP	No	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT does not support an MLPP global diversion number.[8] Diversion to the alternate directory number is provided in lieu of the BNEA.[9] The operational impact is minor.
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: Full compliance of DSN Common Call Features was not met.[11, 12, 13, 14, 15, 16, 17] The operational impact is minor.

Table 2-4. SUT Interoperability Test Summary (continued)

Attendant	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
Public Safety	Yes	Certified	All public safety features are conditional. The SUT Met all critical CRs and FRs for Basic 911. The SUT does not support the other public safety features. These are not required features for a PBX 1. There is no risk associated with the SUT not supporting these features.[18]
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Conferencing	Yes	Certified	Meet-Me Conferencing is met through the use of the Cisco MeetingPlace. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.
Nailed-up Connections	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
DSN Hotline Services	No	Not Tested	This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.
MLPP	Yes	Certified	Met all critical CRs and FRs with the following minor exception: The SUT does not support the Loss of Command and Control announcement.[19]
Call Processing	Yes	Certified	Met all critical CRs and FRs.
ISDN Services	Yes	Certified	Met all critical CRs and FRs.[5]
Synchronization	Yes	Certified	Met all critical CRs and FRs.
Reliability	Yes	Certified	Met all critical CRs and FRs.
Security	Yes	See note 20.	See note 20.
VoIP System	No	Certified	The SUT is certified for VoIP specifically with any certified ASLAN posted on the UC APL. See notes 21 and 22.

Network Gateways

Gateway	Interface & Signaling	Critical	Status	Remarks
PSTN	T1 CAS (DTMF, MFR1, DP)	No	Not Certified	T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release. Wink start recognition is not within the required tolerance.[1] An off-hook seizure below the minimum limit is treated as valid.[2] A call fails to complete after trunk preemption.[3] The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
	E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRs with the following minor exception: This interface does not support NFAS.[5] The operational impact is minor.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
	2-Wire Analog Ground Start (GR-506-CORE)	No	Certified	Met all critical CRs and FRs.

Table 2-4. SUT Interoperability Test Summary (continued)

NOTES:

1 T1 CAS wink start recognition is not within the required tolerance of 100 ms to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

2 The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

3 During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

4 ROUTINE calls attempted over a trunk that is broken receive a T120 in lieu of an ICA. Calls above ROUTINE attempted over a trunk that is broken receive a BPA in lieu of an ICA. The operational impact is minor because they are treated with a BPA and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

5 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

6 A discrepancy exists that is associated with the monitoring tool that SUT uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

7 A configuration change was required on the analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

8 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified CallManager. The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

9 When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the BNEA is not provided. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.

10 When an analog station is active with a call and is preempted by a higher precedence call, the analog station receives the proper PNT. However, after going on hook, the station rings at ROUTINE. This was found to be a minor impact because the station is still preempted correctly.

11 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX 1 for all instruments, however this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a PBX 1.

12 The SUT does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

13 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

14 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

15 A short "ping" ring is not provided when calls are forwarded; however, the phone does visually display that call forward variable is enabled. There is a minor operational impact.

16 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. DISA adjudicated this anomaly as having a minor operational impact because the preempted user receives PNT and the other members remain connected.

Table 2-4. SUT Interoperability Test Summary (continued)

NOTES (continued):

17 When a ROUTINE call is placed to a hunt group, and a ring-no-answer condition occurs, the calling party is diverted to the MLPP alternate directory number. This configuration must be done to allow correct treatment to be provided to precedence calls above ROUTINE that are placed to the hunt group. The UCR requires this only for Precedence above ROUTINE calls. There is no operational impact.

18 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

19 The SUT does not support the Loss of C2 announcement. This announcement is invoked only when a DSN subscriber is automatically routed to a non-MLPP network. DISA adjudicated this anomaly as having a minor operational impact because this announcement would rarely be invoked on a PBX 1. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change the Loss of C2 announcement from required to conditional for a PBX 1.

20 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

21 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor Letter of Compliance signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the Department of Defense Information Technology Standards Registry (DISR).
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

22 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.
b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.
c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP Phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

Table 2-4. SUT Interoperability Test Summary (continued)

LEGEND:

ANSI	American National Standards Institute	LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
APL	Approved Products List	Mbps	Megabits per second
ASLAN	Assured Services Local Area Network	MFR1	Multi-Frequency Recommendation 1
BNEA	Busy Not Equipped Announcement	MLPP	Multi-Level Precedence and Preemption
BPA	Block Precedence Announcement	ms	milliseconds
BRI	Basic Rate Interface	NI 1/2	National ISDN Standard 1 or 2
C2	Command and Control	NI2	National ISDN Standard 2
CAS	Channel Associated Signaling	NFAS	Non Facility Associated Signaling
CRs	Capability Requirements	PBX 1	Private Branch Exchange 1
DISA	Defense Information Systems Agency	PMO	Program Management Office
DP	Dial Pulse	PNT	Preemption Notification Tone
DSN	Defense Switched Network	PRI	Primary Rate Interface
DSS1	Digital Subscriber Signaling 1	PSTN	Public Switched Telephone Network
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.955.3	ISDN Signaling standard for E1 MLPP
FRs	Feature Requirements	SS7	Signaling System 7
GR	Generic Requirement	SUT	System Under Test
GR-506-CORE	LSSGR: Signaling for Analog Interfaces	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ICA	Isolated Code Announcement	T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
IPv4	Internet Protocol version 4	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
IPv6	Internet Protocol version 6	UC	Unified Capabilities
ISDN	Integrated Services Digital Network	UCR	Unified Capabilities Requirements
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector	VoIP	Voice over Internet Protocol
JITC	Joint Interoperability Test Command		

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi.

Table 2-5. SUT Interoperability Requirements/Status

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP)	No	Not Tested (See note 1.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Not tested	
				Line Signaling (R)	UCR Section 5.2	Not tested	
				Normal Wink Start Operations (C)	UCR Section 5.3.3.1.1	Not tested	See note 2.
				Glare Operation (C)	UCR Section 5.3.3.1.2	Not tested	
				Abnormal Wink Start (C)	UCR Section 5.3.3.2.1	Not tested	See note 2.
				Glare Resolution (C)	UCR Section 5.3.3.2.2	Not tested	
				Call for Service Timing (R)	UCR Section 5.3.5	Not tested	
				Guard Timing (R)	UCR Section 5.3.6	Not tested	
				Satellite Timing (C)	UCR Section 5.3.7	Not tested	
				Disconnect Control (C)	UCR Section 5.3.8	Not tested	
				Reselect and Retrial (C)	UCR Section 5.3.9	Not tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.3.10	Not tested	See note 3.
				Dial-Pulse Signals (C)	UCR Section 5.4.1	Not tested	
				DTMF Signaling (C)	UCR Section 5.4.2	Not tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.4.2.1	Not tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.4.3	Not tested	
				Alerting Signals and Tones (R)	UCR Section 5.5	Not tested	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.7.1.4	Not tested	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.7.1.4.2	Not tested	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.7.1.4.3	Not tested	
				Message Functional Definition and Content (R)	UCR Section 5.7.1.4.4	Not tested	
				General Message Format and Information Elements Coding (R)	UCR Section 5.7.1.4.5	Not tested	
				PCM-24 Digital Trunk Interface (R)	UCR Section 7.1	Not tested	
				Interface Characteristics (R)	UCR Section 7.1.1	Not tested	
				Supervisory Channel Associated Signaling (C)	UCR Section 7.1.2	Not tested	
				Clear Channel Capability (R)	UCR Section 7.1.3	Not tested	
				Alarm and Restoral Requirements (R)	UCR Section 7.1.4	Not tested	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 7.3	Not tested	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not tested	
				Trunk Group-Remove from Service (C)	UCR Section 2.5.5	Not tested	
				Trunk Group-Restore to Service (C)	UCR Section 2.5.6	Not tested	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 CAS (MFR1, DTMF, DP) (continued)	No	Not Tested (See note 1.)	Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 CAS (MFR1, DTMF, DP)	No (Europe only)	Not Tested (See note 4.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Not Tested	
				Line Signaling (R)	UCR Section 5.2	Not Tested	
				Normal Wink Start Operations (C)	UCR Section 5.3.3.1.1	Not Tested	
				Glare Operation (C)	UCR Section 5.3.3.1.2	Not Tested	
				Wink Start (C)	UCR Section 5.3.3.2.1	Not Tested	
				Glare Resolution (C)	UCR Section 5.3.3.2.2	Not Tested	
				Call for Service Timing (R)	UCR Section 5.3.5	Not Tested	
				Guard Timing (R)	UCR Section 5.3.6	Not Tested	
				Satellite Timing (C)	UCR Section 5.3.7	Not Tested	
				Disconnect Control (C)	UCR Section 5.3.8	Not Tested	
				Reselect and Retrial (C)	UCR Section 5.3.9	Not Tested	
				Off-Hook Supervision Transition (C)	UCR Section 5.3.10	Not Tested	
				Dial-Pulse Signals (C)	UCR Section 5.4.1	Not Tested	
				DTMF Signaling (C)	UCR Section 5.4.2	Not Tested	
				Standard Digit Format for Precedence (C)	UCR Section 5.4.2.1	Not Tested	
				MFR1 2/6 Signaling (C)	UCR Section 5.4.3	Not Tested	
				Alerting Signals and Tones (R)	UCR Section 5.5	Not Tested	
				PCM-30 Digital Trunk Interface (C)	UCR Section 7.2	Not Tested	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 7.3	Not Tested	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	
				Trunk Group-Remove from Service (C)	UCR Section 2.5.5	Not Tested	
				Trunk Group-Restore to Service (C)	UCR Section 2.5.6	Not Tested	
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	
				Secure calls (R)	CJCSI 6215.01C	Not Tested	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a)	Yes	Certified	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				National ISDN 1/2 Primary Access (R)	UCR Section 2.3.4.1	Met	See note 5.
				ISDN ANSI MLPP Service Capability (R)	UCR Section 2.3.4.1.1	Met	
				Call for Service Timing (R)	UCR Section 5.3.5	Met	
				Disconnect Control (C)	UCR Section 5.3.8	Met	
				Off-Hook Supervision Transition (C)	UCR Section 5.3.10	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.7.1	Met	
				Application (R)	UCR Section 5.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.7.1.4.6	Not Tested	See note 6.
				PCM-24 Digital Trunk Interface (R)	UCR Section 7.1	Met	
				Interface Characteristics (R)	UCR Section 7.1.1	Met	
				Clear Channel Capability (R)	UCR Section 7.1.3	Met	
				Alarm and Restoral Requirements (R)	UCR Section 7.1.4	Met	See note 7.
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 7.3	Met	
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Met	
				Trunk Group-Remove from Service (C)	UCR Section 2.5.5	Not Tested	See note 6.
				Trunk Group-Restore to Service (C)	UCR Section 2.5.6	Not Tested	See note 6.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
T1 ISDN PRI NI 1/2 (ANSI T1.619a) (continued)	Yes	Certified	Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 6.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3)	No (Europe only)	Not certified (See note 8.)	Trunking	Direct Inward Dialing (C)	UCR Section 2.3.2	Met	
				ITU-T ISDN Primary Access (C)	UCR Section 2.3.4.2	Met	
				ITU-T ISDN Primary Access Digital Subscriber Signaling System Number 1 MLPP (C)	UCR Section 2.3.4.2.1	Not Tested	See note 6.
				Call for Service Timing (R)	UCR Section 5.3.5	Not Tested	See note 6.
				Disconnect Control (C)	UCR Section 5.3.8	Met	
				Off-Hook Supervision Transition (C)	UCR Section 5.3.10	Met	
				DSN ISDN User-to-Network Signaling (R)	UCR Section 5.7.1	Met	
				Application (R)	UCR Section 5.7.1.1	Met	
				Physical Layer (R)	UCR Section 5.7.1.2	Met	
				Data Link Layer (R)	UCR Section 5.7.1.3	Met	
				Data Link Connection (R)	UCR Section 5.7.1.3.1	Met	
				Peer-to-Peer Procedures of Data-Link Layer (R)	UCR Section 5.7.1.3.2	Met	
				Layer 3 DSN User-to-Network Signaling (R)	UCR Section 5.7.1.4	Met	
				DSN User-to-Network Signaling for Circuit-Switched Bearer Services (R)	UCR Section 5.7.1.4.2	Met	
				Sequence of Messages for DSN Circuit-Switched Calls (R)	UCR Section 5.7.1.4.3	Met	
				Message Functional Definition and Content (R)	UCR Section 5.7.1.4.4	Met	
				General Message Format and Information Elements Coding (R)	UCR Section 5.7.1.4.5	Met	
				Supplementary Services (C)	UCR Section 5.7.1.4.6	Not Tested	See note 6.
				PCM-30 Digital Trunk Interface (C)	UCR Section 7.2	Met	
				Interoperation of PCM-24 and PCM-30 (C)	UCR Section 7.3	Not Tested	See note 6.
				Integrated Digital Loop Carrier (C)	UCR Section 7.5	Not Tested	See note 6.
				Trunk Group-Remove from Service (C)	UCR Section 2.5.5	Not Tested	See note 6.
				Trunk Group-Restore to Service (C)	UCR Section 2.5.6	Not Tested	See note 6.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Trunk Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
E1 ISDN PRI (ITU-T Q.955.3) (continued)	No (Europe only)	Not Certified (See note 8.)	Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				56 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				64 kbps switched data (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				NX56 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				NX64 synchronous BER (R: PRI only)	UCR Section 3.10	Not Tested	See note 6.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
			VTC	ITU-T H.320 (R: PRI only)	FTR 1080B-2002	Not Tested	See note 6.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Line Interfaces							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
2-Wire Loop Start Analog	Yes	Certified	Access	Directory Number Identification (R)	UCR Section 2.1.1	Met	
				PBX Line (C)	UCR Section 2.3.1	Met	
				Analog Line (R)	UCR Section 2.3.5	Met	
				Basic Line Test Capabilities (R)	UCR Section 2.5.4.1.1	Met	
				Advanced Line Test Capabilities (C)	UCR Section 2.5.4.1.2	Not Tested	See note 6.
				Loop Start Line (R: 2-Wire Analog only)	UCR Section 5.2.1	Met	
				Reverse Battery (R)	UCR Section 5.3.1	Met	
				Alerting Signals and Tones (R)	UCR Section 5.5	Met	See note 9.
			Voice	MOS (R)	CJCSI 6215.01C	Met	
				Secure calls (R)	CJCSI 6215.01C	Met	See note 10.
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Met	
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Met	
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Met	
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	Access	Directory Number Identification (R)	UCR Section 2.1.1	Not Tested	See note 11.
				National ISDN 1/2 Basic Access (C)	UCR Section 2.3.3	Not Tested	See note 11.
				Alerting Signals and Tones (R)	UCR Section 5.5	Not Tested	See note 11.
				S/T Reference Point (R)	UCR Section 5.7.1.2.1	Not Tested	See note 11.
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	See note 11.
				Secure calls (R)	CJCSI 6215.01C	Not Tested	See note 11.
			Facsimile	Analog: ITU-T T.4 (R)	DISR	Not Tested	See note 11.
			Data	Modem (VBD) (R)	CJCSI 6215.01C	Not Tested	See note 11.
				Secure data (STE/STU-III) (R)	CJCSI 6215.01C	Not Tested	See note 11.
			VTC	ITU-T H.320 (R: BRI only)	FTR 1080B-2002	Not Tested	See note 11.
2-Wire Proprietary Digital	No	Not Tested	Access	Directory Number Identification (R)	UCR Section 2.1.1	Not Tested	See note 11.
				Alerting Signals and Tones (R)	UCR Section 5.5	Not Tested	See note 11.
			Voice	MOS (R)	CJCSI 6215.01C	Not Tested	See note 11.
				Secure calls (R)	CJCSI 6215.01C	Not Tested	See note 11.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Common Features	Yes	Certified	Individual Lines (R)	UCR Section 2.1	Met	
			Denied originating service (C)	UCR Section 2.1.3	Not Tested	See note 12.
			Code restriction and diversion (C)	UCR Section 2.1.4	Met	
			Call waiting (R)	UCR Section 2.1.5	Met	See notes 12 and 13.
			Three-way calling (R)	UCR Section 2.1.6	Met	See notes 12, 14, and 15.
			Add-on transfer, conference calling, and call hold (C)	UCR Section 2.1.7	Met	
			Call Transfer Individual – All calls (R)	UCR Section 2.1.7.1	Met	See notes 12 and 15.
			Call Transfer - Internal Only (R)	UCR Section 2.1.7.2	Met	See notes 12 and 15.
			Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (R)	UCR Section 2.1.7.3	Met	See notes 12 and 15.
			Call Transfer – Outside (R)	UCR Section 2.1.7.4	Met	See notes 12 and 15.
			Call Transfer – Add-On Restricted Station (C)	UCR Section 2.1.7.5	Not Tested	See note 6.
			Call Transfer – Attendant (C)	UCR Section 2.1.7.6	Not Tested	See note 6.
			Call Hold (R)	UCR Section 2.1.7.7	Met	See notes 12 and 15.
			Conference Calling – Six Way Station Controlled (C)	UCR Section 2.1.7.8	Met	
			Call forwarding Variable (R)	UCR Section 2.1.8.1	Met	See notes 12 and 15.
			Call Forward Busy Line (R)	UCR Section 2.1.8.2	Met	See notes 12 and 15.
			Call Forwarding – Don't Answer – All Calls (R)	UCR Section 2.1.8.3	Met	See notes 12 and 15.
			Selective Call Forwarding (C)	UCR Section 2.1.8.4	Met	
			Call pick-up (C)	UCR Section 2.1.9	Met	See notes 12 and 15.
			Address Translation (C)	UCR Section 2.7	Met	
			Assured Dial Tone (C)	UCR Section 2.9	Met	
Attendant	No	Not Tested	Attendant Features (C)	UCR Section 2.2	Not Tested	See note 6.
Public Safety	Yes	Certified	Emergency Service (911) Caller (R)	UCR Section 2.4.1.1	Met	See note 16.
			Emergency Service (911) Public Safety Answering Service (C)	UCR Section 2.4.1.2	Not Tested	See note 16.
			Enhanced Emergency Service (E911) (C)	UCR Section 2.4.1.3	Not Tested	See note 16.
			Trace of terminating calls (C)	UCR Section 2.4.2	Not Tested	See note 16.
			Outgoing call trace (C)	UCR Section 2.4.3	Not Tested	See note 16.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Conferencing	Yes	Certified	Preset Conferencing (C)	UCR Section 2.6	Not Tested	See note 17.
			Meet-Me Conferencing (R)	UCR Section 2.6.2	Met	See note 17.
			Progressive Conferencing (C)	UCR Section 2.6.3	Not Tested	See note 17.
Nailed-up Connections	No	Not Tested	Nailed-Up Connections (C)	UCR Section 2.8	Not Tested	See note 6.
DSN Hotline Services	No	Certified	DSN Analog Hotline Service (C)	UCR Section 2.12	Not Tested	See note 6.
MLPP	Yes	Certified	MLPP Overview (R)	UCR Section 3.1	Met	See notes 18 and 19.
			Preemption in the Network (R)	UCR Section 3.2	Met	
			Network Facility with Lower Precedence Calls (R)	UCR Section 3.2.1	Met	
			Network Facility with Equal or Higher Precedence Calls (R)	UCR Section 3.2.2	Met	
			Precedence Call Diversion (R)	UCR Section 3.3	Met	See note 20.
			Channel Associated Signaling (C)	UCR Section 3.4.1	Not Tested	See notes 1 and 21.
			Primary Rate Interface (R)	UCR Section 3.4.2	Met	
			Analog Line MLPP (R)	UCR Section 3.5	Met	
			ISDN MLPP Basic Rate Interface (C)	UCR Section 3.6	Not Tested	See note 11.
			ISDN Primary Rate Interface (R)	UCR Section 3.7	Met	
			Precedence Call Waiting (R)	UCR Section 3.8.1	Met	See note 22.
			Call Forwarding (R)	UCR Section 3.8.2	Met	See note 12.
			Call Transfer (R)	UCR Section 3.8.3	Met	See note 12.
			Call Hold (R)	UCR Section 3.8.4	Met	See note 12.
			Three-Way Calling (R)	UCR Section 3.8.5	Met	See note 12.
			Call Pickup (C)	UCR Section 3.8.6	Met	See note 12.
			Conferencing (C)	UCR Section 3.8.7	Met	See note 17.
			Multiline Hunt Group (C)	UCR Section 3.8.8	Met	See notes 12 and 15.
			Community of Interest (C)	UCR Section 3.8.9	Not Tested	See note 6.
			MLPP Interaction with EKTS features (C)	UCR Section 3.11	Not Tested	See note 6.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Call Processing	Yes	Certified	Call Treatments (R)	UCR Section 4.1	Met	
			Primary and Alternate Routing (C)	UCR Section 4.2	Met	
			E&M Lead Signaling States (C)	UCR Section 4.3.1	Not Tested	See note 11.
			4-Wire Analog User Access Lines (C)	UCR Section 4.3.2	Not Tested	See note 11.
			2-Wire User Access Lines (R)	UCR Section 4.3.3	Met	
			Termination of Analog Lines (R)	UCR Section 4.3.4	Met	
			DSN User Dialing (R)	UCR Section 4.5.1.1	Met	
			Interswitch and Intraswitch Dialing (R)	UCR Section 4.5.1.2	Met	
			Seven-Digit Dialing (R)	UCR Section 4.5.1.2.1	Met	
			Ten-Digit Dialing (R)	UCR Section 4.5.1.2.2	Met	
			Access Code (R)	UCR Section 4.5.1.3	Met	
			Access Digit (R)	UCR Section 4.5.1.3.1	Met	
			Precedence Digit (R)	UCR Section 4.5.1.3.2	Met	
			Service Digit (R)	UCR Section 4.5.1.3.3	Met	
			Route Code (R)	UCR Section 4.5.1.4	Met	
			Area Code (R)	UCR Section 4.5.1.5	Met	
			Switch Code (R)	UCR Section 4.5.1.6	Met	
			Line Number (R)	UCR Section 4.5.1.7	Met	
			Calling Name Delivery (C)	UCR Section 4.5.1.8.1	Not Tested	See note 6.
			Calling Number Delivery (R)	UCR Section 4.5.1.8.2	Met	
			Emergency Service 911 Conflict Resolution (R)	UCR Section 4.5.1.9	Met	
			DSN Switch Outpulsing Digit Formats (C)	UCR Section 4.5.2	Not Tested	See note 1.
			Standard Directory Number (R)	UCR Section 4.5.3	Met	
			Standard Test Numbers (C)	UCR Section 4.5.4	Not Tested	See note 6.
			Base Services – Abbreviated Numbers (C)	UCR Section 4.5.5	Not Tested	See note 6.
			Digit Reception Requirements (R)	UCR Section 4.5.6	Met	
			Screening (C)	UCR Section 4.5.8	Met	
ISDN Services	Yes	Certified	BRI Access, Call Control and Signaling (C)	UCR Section 10, Table 10-1	Not Tested	See note 11.
			Uniform Interface Configuration for BRIs (C)	UCR Section 10, Table 10-2	Not Tested	See note 11.
			Electronic Key Telephone Systems (EKTS) (C)	UCR Section 10, Table 10-3	Not Tested	See note 6.
			PRI Access, Call Control and Signaling (R)	UCR Section 10, Table 10-4	Met	See note 5.
			PRI Features (R)	UCR Section 10, Table 10-5	Met	See note 5.
			Packet Data Features and Capabilities (C)	UCR Section 10, Table 10-6	Not Tested	See note 11.

Table 2-5. SUT Interoperability Requirements/Status (continued)

DSN Features and Capabilities						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
Synchroniz-ation	Yes	Certified	Line timing mode (R)	UCR Section 11.1.1.2	Met	
			Internal Stratum 4 (R)	UCR Section 11.1.2.2	Met	
			Synchronization Performance Monitoring Criteria (C)	UCR Section 11.2	Met	
			DS1 Traffic Interfaces (C)	UCR Section 11.3	Not Tested	See note 11.
			DS0 Traffic Interconnects (C)	UCR Section 11.4	Not Tested	See note 11.
Reliability	Yes	Certified	System Availability (R)	UCR Section 12.2	Met	
			Backup Power (R)	UCR Section 12.3	Not Tested	See note 23.
			Power Components (R)	UCR Section 12.3.1	Not Tested	See note 23.
			UPS Requirements (R)	UCR Section 12.3.2	Not Tested	See note 23.
			UPS PBX 1 Load Capacity (R)	UCR Section 12.3.2.2	Not Tested	See note 23.
			Backup Power (Environmental) (R)	UCR Section 12.3.3	Not Tested	See note 23.
			Alarms (R)	UCR Section 12.3.4	Not Tested	See note 23.
Security	Yes	Certified	GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	UCR Section 13	Met	See note 24.
VoIP						
Feature/ Capability	**Critical**	**Feature Status**	**UCR Requirement**	**Reference**	**Test Results**	**Remarks**
VoIP System	No	Certified (See note 25.)	Voice Quality with MOS of 4.0 or better (R)	UCR App. 3, para. A3.2.1	Met	
			ITU-T G.711 PCM CODEC (R)	UCR App. 3, para. A3.2.2	Met	
			MLPP (R)	UCR App. 3, para. A3.2.3	Met	
			Security (R)	UCR App. 3, para. A3.2.4	Met	
			Network management (C)	UCR App. 3, para. A3.2.5	Met	
			System timing (R)	UCR App. 3, para. A3.2.6	Met	
			Latency ≤ 60 milliseconds (R)	UCR App. 3, para. A3.2.7	Met	
			IPv6 capable (R)	UCR App. 3, para. A3.2.8	Met	See note 26.
			Service Class Tagging (R)	UCR App. 3, para. A3.2.9	Met	
			VoIP System Downtime (IP network 80 min/yr Subscriber 20 min/yr)	UCR App. 3, para. A3.2.10	Met	

Table 2-5. SUT Interoperability Requirements/Status (continued)

Network Gateways							
Interface	**Critical**	**Interface Status**	**UCR Requirement**		**Reference**	**Test Results**	**Remarks**
PSTN (See note 27.)	No	Certified	Trunking	Positive Identification Control (C)	CJCSI 6215.01C	Met	
				On-Netting (C)	CJCSI 6215.01C	Met	
				Off-Netting (C)	CJCSI 6215.01C	Met	
				Ground Start Line (R)	UCR Section 5.2.2	Met	
				Immediate Start (C)	UCR Section 5.3.2	Met	
				Delay Dial (C)	UCR Section 5.3.4	Met	

NOTES:

1 T1 CAS is supported by the SUT; however, it was not tested with this software release because critical interoperability discrepancies were discovered during testing of a previous software release that were not fixed by the vendor. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

2 T1 CAS wink start recognition is not within the required tolerance of 100 ms to 350 ms. The SUT will only recognize a wink from 140 ms to 280 ms. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

3 The SUT will treat any off-hook condition (ABCD Channel Associated Signaling bits high) of 12 ms or greater as a valid off-hook seizure and respond with a wink. In accordance with the requirements, signals that are less than 60 ms should be considered invalid. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

4 E1 CAS is supported by the SUT; however it was not tested. The SUT E1 CAS interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

5 The SUT does not support NFAS on their ISDN PRI NI2 interface. DISA's adjudication of this discrepancy was completed on 17 December 2008 and was ruled to have a minor operational impact. Furthermore, DISA, in coordination with the Joint Staff, stated their intent to modify the next update of the UCR to change NFAS for a PBX 1 from required to conditional.

6 This feature is not supported by the SUT. This is not a required feature for a PBX 1. There is no risk associated with the SUT not supporting this feature.

7 A discrepancy exists that is associated with the monitoring tool that SUT uses to check the status of the ISDN PRI trunks on the gateway. The monitoring tool occasionally provides an inaccurate representation of the status of the channels on the trunks when they are busied by the remote switching system. The SUT will occasionally provide an indication that the channel that was busied out by the far-end switch remains in an idle condition. This anomaly can be eliminated by insuring the trunks are busied at both the remote end and at the SUT. Furthermore, when this anomaly does occur, the correct busy state of the trunks is reflected in layer 3 protocol of the ISDN PRI interface, therefore, the operational impact is minor.

8 The SUT E1 ISDN PRI interface is supported by the SUT; however, it does not support ITU-T Q.955.3 MLPP. The SUT E1 ISDN PRI interface is therefore not certified by JITC, or authorized for use by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1. However, this interface is supported as a PSTN interface only.

9 When an analog station is active with a call and is preempted by a higher precedence call, the analog station receives the proper PNT. However, after going on hook, the station rings at ROUTINE. This was found to be a minor impact because the station is still preempted correctly.

10 A configuration change was required on the analog gateways to meet the requirement for interoperability with secure devices, specifically the L3 Omni Secure Wireline Terminal. On the individual voice ports, the minimum and maximum settings for "timing hookflash in" had to be changed to a maximum value of 500 ms and a minimum value of 150 ms. Otherwise, a call that is placed between two Omni devices on the SUT will not disconnect when placed on hook.

11 This interface is not supported by the SUT. This is not a required interface for a PBX 1. There is no risk associated with the SUT not supporting this interface.

12 Call Forward No Answer, Call Forward Busy, and Multi-Line Hunt Service are supported on both VoIP and analog stations. Call Forward Variable, Three-way Calling, Call Hold, Call Pick-up, and Call Transfer are supported on VoIP stations only. These features are required for a PBX1 for all instruments, however this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. Denied Originating Service is not supported by the SUT and is therefore not covered in this certification. This feature is not required for a PBX 1.

13 The SUT does not support Call Waiting. However, there is no operational impact because the requirement is satisfied with multiple line appearances having a busy trigger. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability.

Table 2-5. SUT Interoperability Requirements/Status (continued)

NOTES (continued):

14 A conference disconnect tone is not provided when a three-way conference originator is preempted. This only occurs when an analog station originates the first call. DISA adjudicated this anomaly as having a minor operational impact because the preempted user receives PNT and the other members remain connected.

15 All of the features on the VoIP phones were tested using multiple line appearances. A minimum of two line appearances is required to meet the MLPP interoperability requirements for Call Features with the exception of call hold, call pickup, and call forwarding functions.

16 The SUT only supports emergency service 911 public safety features. The following public safety features are not supported and therefore are not covered in this certification: Trace of terminating calls, Outgoing call trace, Tandem call trace, and Trace of a call in progress. There is no operational impact because these public safety features are not required for a PBX 1.

17 Meet-Me Conferencing is met through the use of an optional adjunct conferencing system call the Cisco MeetingPlace which is covered under a separate certification. The SUT does not support Preset Conferencing or Progressive Conferencing. These features are not required for a PBX 1. There is no risk associated with the SUT not supporting these features.

18 ROUTINE calls attempted over a trunk that is broken receive a T120 in lieu of an ICA. Calls above ROUTINE attempted over a trunk that is broken receive a BPA in lieu of an ICA. The operational impact is minor because they are treated with a BPA and since a PBX 1 cannot support special command and control users, the operational impact is mitigated.

19 When a station classmarked by the SUT as non-preemptable is active with a call and a higher precedence call attempts to directly preempt it, the BNEA is not provided. The operational impact is minor because the call is forwarded to the MLPP alternate directory number that is specified in the station's configuration.

20 The SUT does not support an MLPP global diversion number. Each station must be individually configured with a precedence diversion number from a single location using the Bulk Administration Tool provided with the Cisco Unified CallManager . The operational impact is minor because diversion settings can be configured for all of the stations provisioned on the switch from a single location.

21 During a trunk preemption test over the T1 CAS from the far-end to the SUT, after the preemption occurred the call would fail to complete and no treatment was provided to the call originator. This is a critical requirement for T1 CAS. The SUT T1 CAS interface is not certified by JITC or authorized by the DSN PMO for use within the DSN. This is not a required interface for a PBX 1.

22 Although the SUT does not support Precedence Call Waiting, they do support multiple call appearances on their VoIP stations. This provides the ability for a user to receive additional calls while active with another call. Also, this is a new UCR requirement and the vendor has 18 months (until July 2009) to develop this capability. There is no operational impact.

23 This requirement is a non-testable requirement. It is the responsibility of the respective base/post/camp/station communications agency to provide this with the SUT when installed.

24 Security is tested by DISA-led Information Assurance test teams and published in a separate report, reference (c).

25 An IPv6 capable system or product, as defined in the UCR, paragraph 1.7, shall be capable of receiving, processing, and forwarding IPv6 packets and/or interfacing with other systems and protocols in a manner similar to that of IPv4. IPv6 capability is currently satisfied by a vendor LoC signed by the Vice President of the company. The vendor stated, in writing, compliance to the following criteria:

a. Conformant with IPv6 standards profile contained in the DISR.
b. Maintaining interoperability in heterogeneous environments and with IPv4.
c. Commitment to upgrade as the IPv6 standard evolves.
d. Availability of contractor/vendor IPv6 technical support.

26 All of the SUT components covered under this certification met the IPv6 criteria with the exception of the Cisco 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP-7960G. The 3745, 3725, CP-7940G, and CP7960G do not meet the critical IPv6 capability requirement in accordance with the UCR, paragraph 1.7. However, components that are not currently IPv6 capable and have been identified by the vendor as having no migration path to IPv6, may be certified if the following criteria is met:

a. The component must already be JITC certified and currently fielded within the DSN.
b. This component will not be purchased to be used within the DSN. This component is covered under this certification specifically for software upgrades to existing components.
c. There must be a certified, IPv6-capable component available for replacement. To meet this requirement Cisco has designated the 3845 and 3825 respectively as replacements for the 3745 and 3725 Multiservice Access Routers. Cisco has designated the CP-7941G and CP-7961G IP Phones respectively as replacements for the CP-7940G and CP-7960G IP phones.

27 Voice, facsimile, data, and VTC service requirements for PSTN are identical to DSN with the exception of MLPP.

Table 2-5. SUT Interoperability Requirements/Status (continued)

LEGEND:

ANSI	American National Standards Institute	G.711	PCM of voice frequencies	PBX 1	Private Branch Exchange 1
App.	Appendix	GR	Generic Requirement	PCM	Pulse Code Modulation
BER	Bit Error Ratio	GR-815	Generic Requirements For Network	PCM-24	Pulse Code Modulation - 24 Channels
BNEA	Busy Not Equipped Announcement		Element/Network System (NE/NS) Security	PCM-30	Pulse Code Modulation - 30 Channels
BPA	Blocked Precedence Announcement	H.320	Standard for Narrowband VTC	PMO	Program Management Office
BRI	Basic Rate Interface	ICA	Isolated Code Announcement	PNT	Preemption Notification Tone
C	Conditional	IP	Internet Protocol	PRI	Primary Rate Interface
CAS	Channel Associated Signaling	IPv4	Internet Protocol version 4	PSTN	Public Switched Telephone Network
CJCSI	Chairman of the Joint Chiefs of Staff	IPv6	Internet Protocol version 6	Q.955.3	ISDN Signaling Standard for E1 MLPP
	Instruction	ISDN	Integrated Services Digital Network	R	Required
CODEC	Coder/Decoder	IT	Information Technology	S/T	ISDN BRI 4-wire interface
DIACAP	DoD Information Assurance Certification and	ITU-T	International Telecommunication Union -	SS7	Signaling System 7
	Accreditation Process		Telecommunication Standardization Sector	STE	Secure Terminal Equipment
DISA	Defense Information Systems Agency	JITC	Joint Interoperability Test Command	STIGs	Security Technical Implementation Guides
DISR	DoD IT Standards Registry	kbps	kilobits per second	STU-III	Secure Telephone Unit -3rd generation
DoD	Department of Defense	LoC	Letter of Compliance	SUT	System Under Test
DoDI	Department of Defense Instruction	Mbps	Megabits per second	T1	Digital Transmission Link Level 1 (1.544
DP	Dial Pulse	MFR1	Multi-Frequency Recommendation 1		Mbps)
DS0	Digital Signal Level 0 (64 kbps)	min	minute	T1.619a	SS7 and ISDN MLPP Signaling Standard for
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048	MLPP	Multi-Level Precedence and Preemption		T1
	Mbps European)	MOS	Mean Opinion Score	T.4	Standardization of Group 3 facsimile terminals
DSN	Defense Switched Network	ms	millisecond		for document transmission
DTMF	Dual Tone Multi-Frequency	NFAS	Non Facility Associated Signaling	TDM	Time Division Multiplexing
E&M	Ear and Mouth	NI 1/2	National ISDN Standard 1 or 2	UCR	Unified Capabilities Requirements
E1	European Basic Multiplex Rate (2.048 Mbps)	NI2	National ISDN Standard 2	UPS	Uninterruptible Power Supply
EKTS	Electronic Key Telephone System	NX56	Data format restricted to multiples of 56 kbps	VBD	Variable bit data
FTR	Federal Telecommunications	NX64	Data format restricted to multiples of 64 kbps	VoIP	Voice over Internet Protocol
	Recommendation	para.	paragraph	VTC	Video Teleconferencing
FTR 1080B-2002	Video Teleconferencing Services	PBX	Private Branch Exchange	yr	year